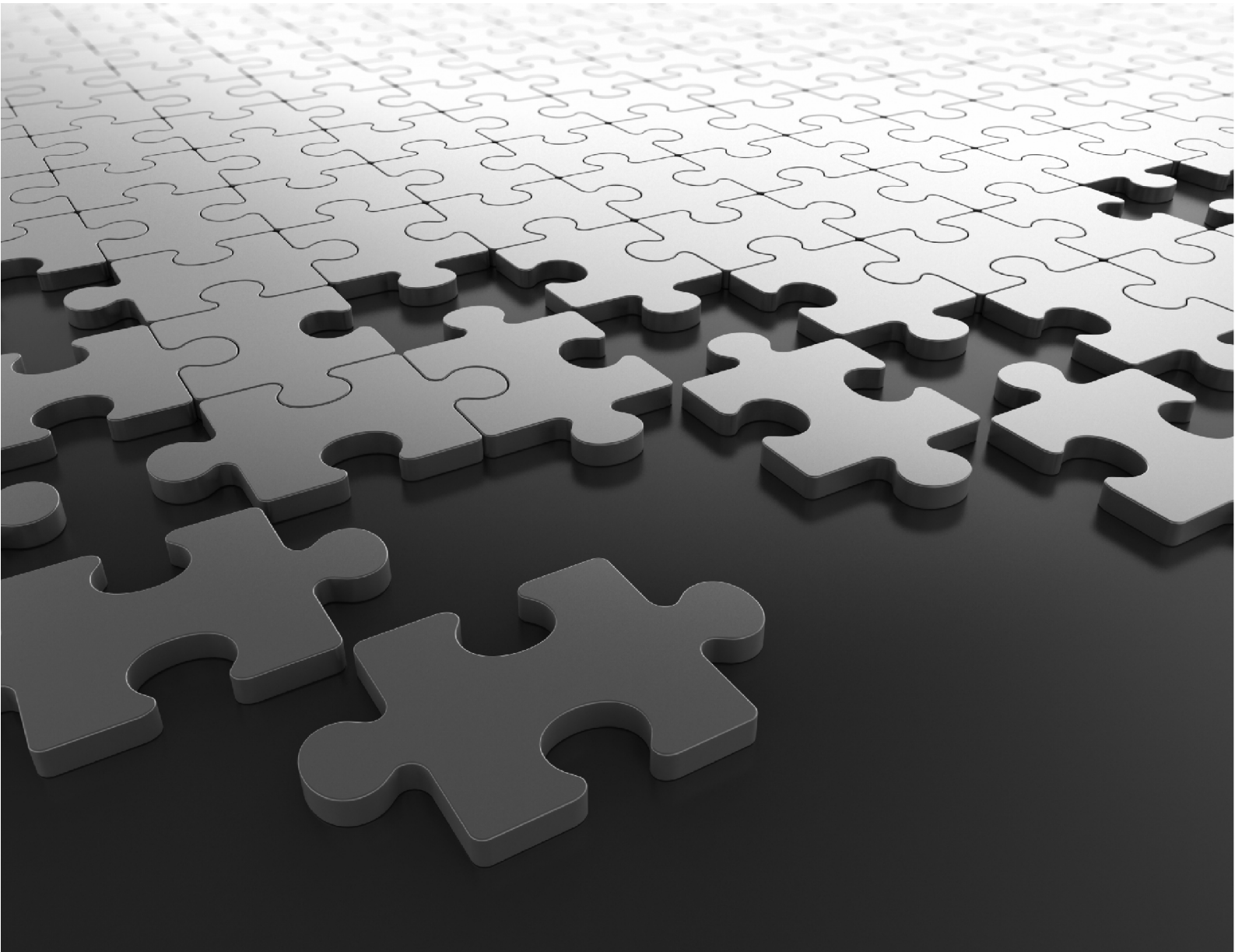


SETU ATA211G
System Manual



SETU ATA211G
VoIP-FXS-GSM Adaptor

System Manual



Documentation Disclaimer

Matrix Comsec reserves the right to change, at any time, without prior notice, the product design, specifications, components, as engineering and manufacturing may warrant.

This is a general documentation for all models/configurations of the product. The product may not support some of the features/facilities described in this document.

While every reasonable effort has been made to ensure accuracy of content in this document, Matrix Comsec assumes no responsibility for errors or omissions that may occur herein. No liability is assumed for damages, costs, expenses resulting from unauthorized modifications or repairs to the product; failure to use information or to comply with the installation, operation and maintenance instructions contained in this document.

Copyright

All rights reserved. No part of this system manual may be copied or reproduced in any form or by any means without the prior written consent of Matrix Comsec.

Version 2

Release date: February 3, 2011



Contents

Introduction.....	1
Welcome	1
About this System Manual	1
How to Read this System Manual	2
Know Your SETU ATA211G.....	5
Overview	5
Applications of SETU ATA211G	8
Getting Started.....	11
Preparing for Installation	11
Connecting SETU ATA211G	13
Configuring SETU ATA211G	16
Configuring Basic Settings.....	21
Region	23
SIP Trunks	25
Mobile Port	38
FXS Port	48
Passwords	56
Network Parameters	59
Advanced Settings	63
System Parameters	64
Called Party Number Table	69
Dialed Number Table	71
Peer-to-Peer Numbers	74
Mobile Emergency Numbers	78
Digest Authentication	80
PIN Authentication	83
Daylight Saving Time	85
Static Routing	88
Reinstate Factory Defaults	91
Restart SETU ATA211G	94
Syslog	95
PCAP	97
Software Upgradation	100
Auto Configuration	101

Features of SETU ATA211G	105
<i>Call Hold</i>	<i>105</i>
<i>Making a Second Call</i>	<i>106</i>
<i>Call Transfer</i>	<i>107</i>
<i>Call Toggle</i>	<i>109</i>
<i>Call Waiting</i>	<i>110</i>
<i>Conference</i>	<i>113</i>
<i>Do-Not-Disturb</i>	<i>114</i>
<i>Call Forward</i>	<i>116</i>
<i>Caller ID Restriction</i>	<i>120</i>
<i>Selective SIP Trunk Access</i>	<i>122</i>
<i>Speed Dialing</i>	<i>123</i>
<i>Supplementary Services of Service Provider</i>	<i>130</i>
<i>Multi Stage Dialing</i>	<i>131</i>
<i>End of Dialing</i>	<i>136</i>
<i>Mobile Network Selection</i>	<i>137</i>
<i>Return Call to Original Caller (RCOC)</i>	<i>139</i>
System Status	143
<i>Viewing System Status from Jeeves</i>	<i>143</i>
<i>System</i>	<i>148</i>
<i>Viewing System Status on Phone</i>	<i>149</i>
Appendix	151
<i>Call Progress Tone Generation</i>	<i>151</i>
<i>Ring Cadences</i>	<i>152</i>
<i>Frequently Asked Questions (FAQs)</i>	<i>153</i>
<i>Acronyms</i>	<i>157</i>
<i>Features at a Glance</i>	<i>159</i>
<i>System Commands</i>	<i>160</i>
<i>Product Specifications</i>	<i>162</i>
<i>Warranty Statement</i>	<i>165</i>
<i>Regulatory Information</i>	<i>167</i>
<i>Open Source Licensing Terms and Conditions</i>	<i>169</i>
Index	177

Welcome

Thank You for choosing SETU ATA211G! This product is designed to give you the highest performance, combined with real ease of use. We hope you will make the optimum use of intelligent, intuitive, feature-packed VoIP Phone Adapter. We recommend you to read this document carefully to get acquainted with the product before installing and operating it.

About this System Manual

This document contains detailed information and instructions for installing, configuring, and operating SETU ATA211G. It also contains information on maintenance and safety instructions.

Intended Audience

This manual is aimed at:

- **Network Administrators and System Engineers**, who will install, maintain and support SETU ATA211G. It is assumed that they have some experience in installing VoIP - GSM Gateways and are familiar with VoIP technology.
- **End Users**, persons/organizations who will use SETU ATA211G. They include home users, personnel of small and medium businesses, large enterprises, other commercial and public organizations/ institutions.

Organization of this document

This System Manual contains following topics:

- **Introduction:** Gives information about this System Manual.
- **Know Your SETU ATA211G:** Provides overview of SETU ATA211G.
- **Getting Started:** Contains information on how to install SETU ATA211G, how to configure the device using the web-based programming tool, Jeeves, and using the phone connected to the device.
- **Configuring Basic Settings:** Provides instructions for configuring the basic parameters of SETU ATA211G, which are sufficient to get the system into operation.

- **Configuring Advance Settings:** Contains instructions for configuring the more advanced features and facilities of SETU ATA211G.
- **Features of SETU ATA211G:** Describes in detail, the features of SETU ATA211G, along with instructions for configuring and using these features.
- **System Status:** Displays the status indicators of system parameters like SIP, Mobile, Network (LAN and WAN), etc and details of the System Firmware.

How to Read this System Manual

This System Manual is organized in a manner to help you get familiar with SETU ATA211G, learn how to install it, how to connect it to the IP network, how to configure the Basic Settings that allow you to start making and receiving calls, and move on to the Advanced configuration to use the special features and functions. You may read it in the sequence it is organized. .

Conventions used in this System Manual

Instructions

The instructions in this document are written in a step-by-step format. Each step, its result, indication/notification, where and as it occurs, has been described.

Notices

The following symbols have been used for notices to draw your attention to important items.



Important: *to indicate something that requires your special attention or to remind you of something you might need to do when you are using the system.*



Tip: *To indicate a helpful hint giving you an alternative way to operate the system or carry out a procedure, or use a feature more efficiently.*

Cross-references

Cross references are given in blue font and hyperlinked. You may click on the hyperlinked text to reach the source of the cross-reference.

Terminology

The technical terms and Acronyms used in this manual are standard terms, commonly used in the telecommunications and data communications industry. Considering the broad group of intended users of this manual, wherever possible, use of jargon is avoided.

Acronyms have been defined in the text and a complete list of Acronyms used in this document is provided in the Appendix.

The words **ATA**, **ATA211G**, **SETU ATA211G** and **System** means SETU ATA211G.

Some of the terms specific to this manual that you will encounter frequently are defined below:

Term	Usage in the document
Admin	A person who install, configure and maintains SETU ATA211G. Admin can also be the user. Admin has full accessibility of the system.
User	A person who uses SETU ATA211G. User has limited access to the system.
Caller/Calling Party	A person who makes calls.
Callee/Called Party	A person who receives the calls.
Transferor	A person who transfers the calls.
Transferee	The remote user whose call is to be transferred (first party) to another remote user (second party)
Transfer Target	The remote user to whom the call is to be transferred (second party).
LAN Administrator	A person who maintains the LAN network of an organization.

Using this manual, you will be able to install, operate and make optimum use of SETU ATA211G. If you encounter any technical problems, please contact your Dealer/reseller or Matrix for support.

Overview

ATA211G is a VoIP Analog Telephone Adapter used as a SIP-FXS-Mobile gateway. It allows you to make and receive voice calls over IP Network as well as PSTN network using conventional telephone instrument.

ATA211G can be used in Residence, SOHO, Corporate Offices, Industries, Workshops, etc. ATA211G can be used to make voice calls to remotely located branch offices of an enterprise using Internet network.

ATA211G also offers flexibility to make VoIP calls over the internet and browse the Internet simultaneously using inbuilt router.

ATA211G uses SIP (Session Initiation Protocol) protocol to make voice calls over the IP network. ATA211G converts the voice traffic into IP packets for transmission over the internet. When a telephone number is dialed by you, ATA211G converts it into an IP call using the Session Initiation Protocol (SIP) and initiates a call, thereby making VoIP calls as easy as normal telephone call. You have flexibility to make a call by selecting any SIP trunk or you can also select Mobile port to make a call.

The number dialed from FXS port can be routed to Mobile port or SIP trunk based on the number dialed. The Incoming call on SIP can be routed to FXS or Mobile Port or to a fixed destination number based on the CLI received. The Incoming call on the Mobile port can be routed to FXS port or SIP trunk as configured in Dialed Number Table.

ATA211G has three voice ports, viz. FXS, SIP and Mobile.

ATA211G has 1 WAN Port, 1 LAN Port, 1 FXS Port, 1 Mobile Port, an antenna connector, a power socket, and 5 LEDs as shown in below figure.

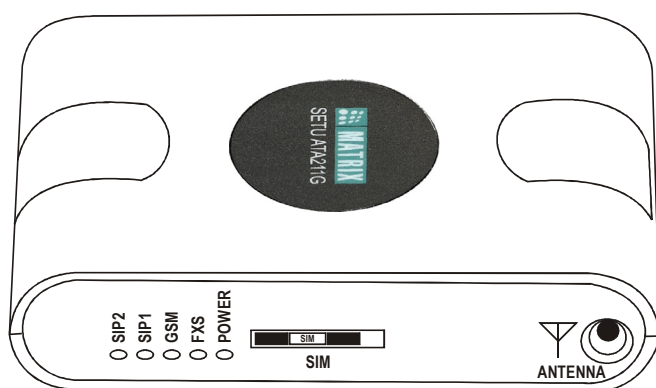


Ports and Connectors

Port Name	Connector	Description
12V DC	DC Jack	Power Adapter
LAN	RJ45	To connect a standalone computer or a LAN Switch.
WAN	RJ45	To connect to the IP network over a DSL Modem or Router or a LAN Switch
FXS	RJ11	To connect a standard telephone instrument or a PBX.
SIM	SIM holder	To connect to GSM network.
Antenna	SMA (Male)	To connect Mobile port antenna.

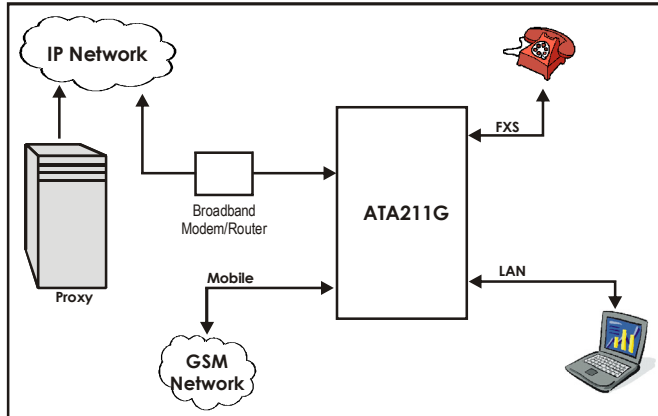
LED

SETU ATA211G has an LED each for the FXS port, the Mobile port, the SIP Trunks (SIP1, SIP2) and POWER, to indicate the status of the ports, and various events occurring on the ports, including errors.

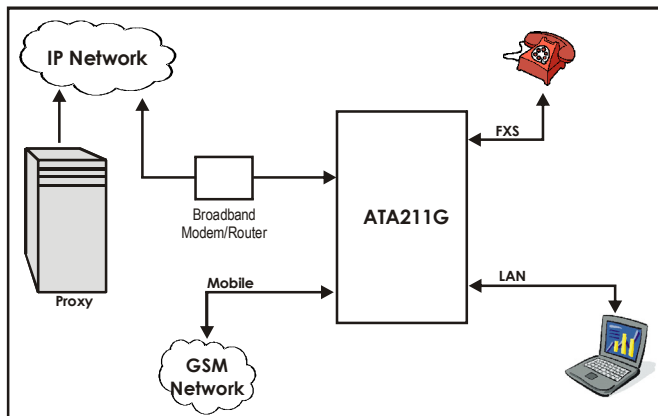


Applications of SETU ATA211G

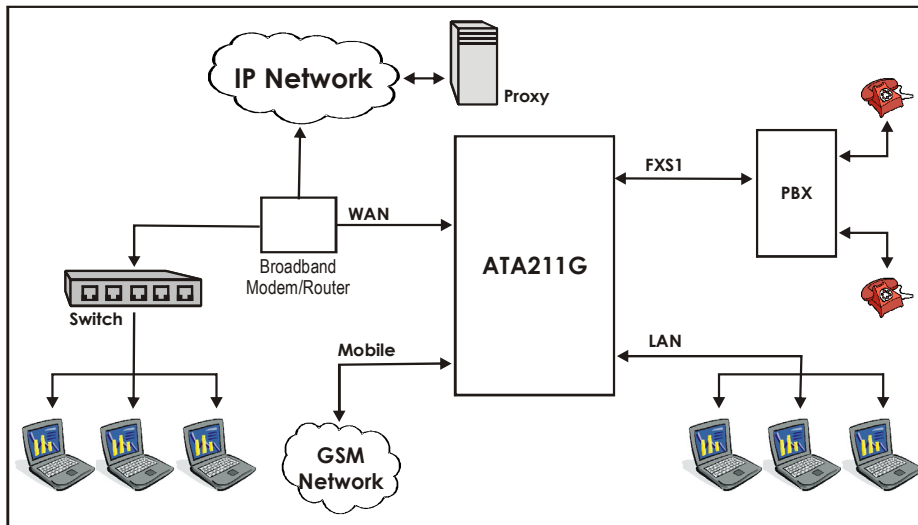
1. **Residential Application:** Home users can use ATA211G to make low-cost international calls to family, friends and relatives.



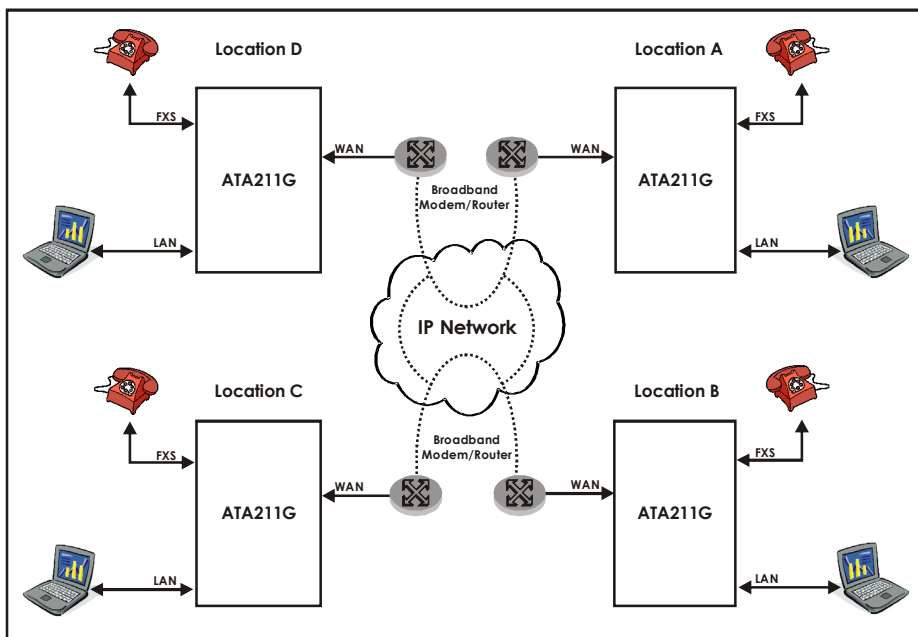
2. **SOHO Application:** Small Office owners can use ATA211G to make low-cost calls to their overseas customers, employees, suppliers, etc.



3. **Business Application:** Corporate Offices can use ATA211G to make low-cost international calls to their overseas customers, employees, suppliers, etc.



4. **Peer-to-Peer Call Application:** Corporate Offices also use SETU ATA211G to communicate between various branch offices, at no cost.



Preparing for Installation

Package Contents



SETU ATA211G
with Antenna



Adaptor 12V, 1.25 Amp
(Country Specific)



Quick Start and User Card



Ethernet Cable (RJ45)



CD containing System Manual,
Quick Start and User Card

- Two Screws M 7/30
- Two Screw Grips
- Warranty Card Set
- External Antenna SMA with Cable

- Unpack SETU ATA211G.
- Verify the package contents. Contact your vendor/Service Provider/administrator, if any of the above listed items is missing or damaged.

Protecting SETU ATA211G and yourself

You are recommended to take the following measures to protect yourself and SETU ATA211G.

Installation Precautions

Do not install the system in a place where,

- it may be exposed to direct sunlight, excessive cold or humidity.
- it may be exposed to dust, water, oil, corrosive fumes.
- it may fall, causing damage to the product.
- shocks or vibrations are frequent or strong.
- there is a water source or body, like a wash basin/kitchen sink, bath-tub, a swimming pool, sprinkler, etc.

Make sure there is enough open space for ventilation at the site of installation.

Safety Instructions

Take every measure to reduce the risk of fire, electric shock and injury to the person handling this product.

- Read all the instructions given in this manual.
- Always switch off the product, unplug the product from the wall outlet when handling it. Removing covers or opening the system or incorrect reassembly increases the risk of electric shock.
- Never open the system in power ON condition.
- Use the adapter provided with the product.
- Make sure telephone cables not touch exposed power line cables.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
- Unplug the product from the wall outlet before cleaning. Do not use liquid cleaners. Use only dry and soft cloth.

Always get your product serviced by a trained, qualified person. Contact your authorised dealer or Matrix support, if:

- liquid has been spilled into the product or the product has been exposed to rain or water.
- the product has been dropped or the cabinet has been damaged.
- the product does not operate normally or exhibits a distinct change in performance.

Warning for RF Safety

The product complies with RF exposure guidelines as per standard FCC 47 CFR part 2. We recommend the following precautions:

- Make sure that the RF Antenna is installed at least 20 cm away from other electronic and radio transmission devices.
- Make sure that the RF Antenna is installed atleast 20 cm away from people's vicinity.
- Do not place magnetic storage media near the product.
- People carrying medical implants like cardiac pacemakers are advised to maintain appropriate distance from the system. They are also advised to avoid being in the vicinity of the product for a long time.
- Do not place the magnetic storage media near the device.
- People carrying medical implants like cardiac pacemakers are advised to maintain appropriate distance from the system. They are also advised to avoid being in the vicinity of the product for a long time.

Mounting the SETU ATA211G

To mount SETU ATA211G on the wall,

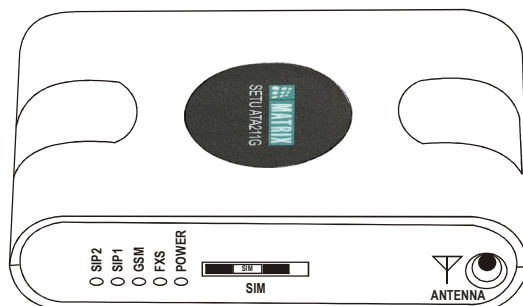
- Select a suitable place.
- Use the mounting template provided to you to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix the screws in the holes, leaving the screw heads protruding a few millimeters from the wall.
- Check the strength of the screws.
- Mount SETU ATA211G on the wall.

Connecting SETU ATA211G



- Connect a standalone computer/ LAN Switch to the **LAN Port** of SETU ATA211G using an Ethernet cable.
- Connect the **WAN Port** of SETU ATA211G to the IP Network - a DSL modem or Router or a LAN Switch - using the Ethernet cable provided to you.
- Connect a telephone to the **FXS Port** of SETU ATA211G, using a standard telephone cable with an RJ11 plug.

You may also connect a PBX to the FXS port of SETU ATA211G.



- Screw the antenna on to the antenna connector.

Protecting the SIM Card

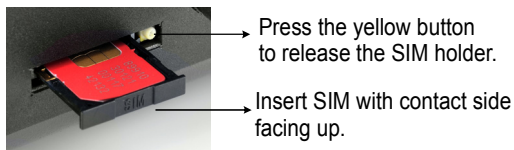
To protect the SIM card, *before* you insert the card in the system, do the following:

- Insert the SIM Card into a Mobile Handset.
- Enable SIM PIN protection.
- Change the SIM PIN to 1234 (default).

- Remove the SIM Card from the Mobile Handset.
- Insert the SIM Card in to the SIM Holder of SETU ATA211G.

You may change the SIM PIN to the desired value after system installation is completed. You can change the SIM PIN from Mobile Port parameters page of Jeeves. See [“Mobile Port”](#) under Basic Settings for instructions.

- Insert SIM Card into the SIM holder of the Mobile Port of SETU ATA211G. To do this,
 - press the SIM holder lock (the small yellow button). You may use a blunt pin to press the lock button.
 - the SIM holder tray emerges from its slot.
 - pull out SIM holder tray from its slot.
 - now, fit the SIM Card on the tray, with its contact side facing up.
 - insert the SIM holder tray back into the slot.



- Connect the power adapter provided with SETU ATA211G to the power jack labeled **12VDC-1A(MAX)**.
- Connect the adapter plug into a power outlet.
- Switch ON power.
- Wait for the Reset Cycle to complete.

Reset Cycle

- **At Power ON,**
 - The Power LED will glow green continuously.
 - All other LEDs - FXS, Mobile, SIP1 and SIP2 will glow Red for ms, where xxx time depends on the timing cycle from boot loader program to init sequence.
 - After few seconds of Power ON, the Mobile LED starts blinking red. It indicates the network availability.
- **On successful completion of initiation cycle,** each LED will glow as per the events summarized in the table below:

Meaning of LEDs

LED	Status	Meaning
FXS	ON	IC Call/OG Call - Speech and Off hook
	Blinks Fast	Error condition during IC/OG call
	Blinks Slow	IC Call: Ring Event OR OG Call: Number being dialed (after the end of dialing)
	OFF	FXS port ON-Hook

LED	Status	Meaning
Mobile	ON	SIM Card is present and or registered with the network.
	Blinking (100 ms ON-100 ms OFF)	SIM Card is absent or de-activated or SIM PIN is faulty or SIM PUK is required
	Blinking (1 ms ON - 1 ms OFF)	SIM Card detected but is not registered with the network
	OFF	ATA211G is powered ON but SIM Card is not detected
SIP1	ON	SIP1 is enabled and is registered with the SIP server or 'Active' status of SIP Trunk.
	Blinks Fast	SIP1 is enabled but not registered with the SIP server (SIP is configured as proxy).
	Blinks Slow	SIP1 is enabled but not registered with the SIP server (SIP is configured for P2P calling).
	OFF	SIP1 is disabled.
SIP2	ON	SIP2 is enabled and is registered with the SIP server or 'Active' status of SIP Trunk.
	Blinks Fast	SIP2 is enabled but not registered with the SIP server (SIP is configured as proxy).
	Blinks Slow	SIP2 is enabled but not registered with the SIP server (SIP is configured for P2P calling).
	OFF	SIP2 is disabled.

You can also view the status of the ports on the *Status* page of the web-based graphical user interface of SETU ATA211G.

Configuring SETU ATA211G

SETU ATA211G provides access to system configuration at two levels: **Admin** and **User**. A distinct set of features and facilities can be configured at each of these levels.

SETU ATA211G can be configured using its Web-based graphic user interface (GUI) and from the phone connected to its FXS port. While the entire system configuration can be done using the GUI, you can configure only certain parameters from the phone.

Admin Mode

Admin is defined as the person who installs, configures and maintains the SETU ATA211G. The Admin can also be the User. In the Admin mode, you have full access to system configuration, all configurable parameters, features and facilities, and system maintenance:

- Basic Settings
- Advanced Settings
- Supplementary Services

Access to Admin mode is protected by a password. The default Admin Password is **1234**. The password may be changed.

User Mode

In the User mode, you can configure specific features and facilities of SETU ATA211G, as listed below:

- Supplementary Features
- Speed Dialing
- Network parameters
- User Password
- Syslog (debug)
- Factory Defaults
- PCAP

Access to the User mode is also protected by a password, referred to as the User Password. The default User Password is **1234**. The password may be changed.

Configuration using Web-based GUI: Jeeves

SETU ATA211G has a graphic user interface (GUI), **Jeeves**, an easy-to-use proprietary software of Matrix for system configuration. Jeeves is an embedded, web server which you can access from the computer you connected to the LAN Port or the WAN Port of SETU ATA211G.

Jeeves allows system configuration in the **Admin** mode and **User** mode.

To access Jeeves, follow these steps:

- Connect a standalone computer to the LAN port of SETU ATA211G. The LAN Port of SETU ATA211G and that of the computer must be in the same Subnet.

The default IP Address of the LAN port of SETU ATA211G is: **192.168.2.21**

The default Subnet Mask of the LAN port of SETU ATA211G is: **255.255.255.0**

If required, change the IP Address and the Subnet of the computer.



You may also connect the LAN port of SETU ATA211G to a LAN switch. If you connect the LAN port of SETU ATA211G to a LAN Switch, make sure that the IP Address of LAN port of SETU does not conflict with the IP Address of any device on the LAN, and SETU ATA211G is in the same Subnet as the LAN computer.

You may change the IP Address and Subnet Mask of the LAN Port of SETU ATA211G by dialing the relevant System Commands from the phone connected to the FXS port of SETU ATA211G. See “” topic.

When installing the system for the first time, you are advised to connect SETU ATA211G to a standalone computer only.

- Make sure a web-browser, either Internet Explorer 7 or Mozilla Firefox, is installed on the computer.
- Open the browser on the computer.
- Type **192.168.2.21**, the default IP address of the LAN port of SETU ATA211G, in the address bar of the browser and press Enter key/click Go.

If you have connected SETU ATA211G to the LAN and have changed the IP Address of the LAN port of SETU ATA211G, enter the *current* IP Address in the address bar of the browser.

The Login page of Jeeves opens.



You can also use the WAN Port of SETU ATA211G to access the embedded web server in the same manner.

*The default IP Address of the WAN Port of SETU ATA211G is: **192.168.1.196***

*The default Subnet Mask of the WAN Port of SETU ATA211G is: **255.255.255.0***

On the Login page of Jeeves, you are offered two Login modes: Admin and User.

To log in as **Admin**,

- In the **Login as** combo box, select **Admin**.
- In the **Password** field, enter **1234**, the default Admin Password.
- Click the **Login** button.
- On successful login, the Admin mode page will open.
- The left pane shows the links **Basic Settings**, **Advanced Settings**, **Supplementary Services**, and **Status**.

Basic Settings are sufficient to get your SETU ATA211G into operation.

- Click the links to expand.

To log in as **User**,

OFFICE : 394-GIDC, Makarpura, Vadodara-390010, India Ph : +91-265-2630555 Fax : +91-265-2636598

WARNING : No part of the system should be copied or reproduced in any form or by any means without the prior written consent of Matrix Comsec Pvt. Ltd.

Copyright © 2010 Matrix Comsec Pvt. Ltd.

This is based on GoAhead Webserver. **WEB SERVER**

- In the **Login as** combo box, select **User**.
- In the **Password** field, enter **1234**, the default User Password.
- Click the **Login** button.
- On successful login, the User mode page will open.
- The left pane displays **User Settings** and **Status** links.
- Click the **User Settings** link to expand. The sub-links to the functions and features that are allowed to be configured from the User mode appear on the left pane: Supplementary Features, Speed Dialing, Network, User Password, Syslog, Factory Defaults and PCAP.
- Click the **Status** link to view the status of the system and the ports.

Configuring using a Phone

You can enter the **Admin** mode and the **User** mode using the phone connected to the FXS Port of SETU ATA211G, and change feature settings, and certain network parameters like the IP Address and Subnet Mask of the LAN and the WAN port of SETU ATA211G, select the Connection Type (IP Addressing) by dialing “[System Commands](#)” from the phone connected to its FXS port of SETU ATA211G.

You can also dial System Commands to view the current IP Addresses of the WAN and LAN Ports (if your phone has a display).

Follow the instructions given below to configure the system from the phone.

Admin Mode

- Lift the receiver of the phone.
- Dial **#19** to enter the Admin mode.
- Dial the Admin Password (default: **1234**).
You will get programming tone.
- Dial the System Command for the feature/facility you want to configure.

For example:

To change IP Address of the LAN Port, dial **12<LAN IP Address>#***.

For example, to change the IP address to 192.168.2.21, dial **12192168002021#***

- When you dial a valid command, SETU ATA211G plays a confirmation tone for 3 seconds, and plays the programming tone.
- Dial **0** to exit configuration mode.

When network parameters are changed, SETU ATA211G will restart, without dial tone/confirmation tone.

You can also view the Network parameters (IP Address, Subnet Mask, Connection Type. etc.) on the phone's display. See “[System Status](#)”.

For a complete list of commands, refer “[System Commands](#)” in the Appendix.

User Mode

- Lift the receiver of the phone.
- Dial **#18** to enter the User mode.
- Dial the User Password (default: **1234**).
- You will get programming tone.
- Dial the command for the feature/facility you want to configure.

For example,

The command for configuring the destination number for Call Forward -Unconditional is: **51-Destination Number-#***

To configure 2001 as the destination number for Call Forward-Unconditional, dial **512001#***

- Dial **0** to exit programming mode.

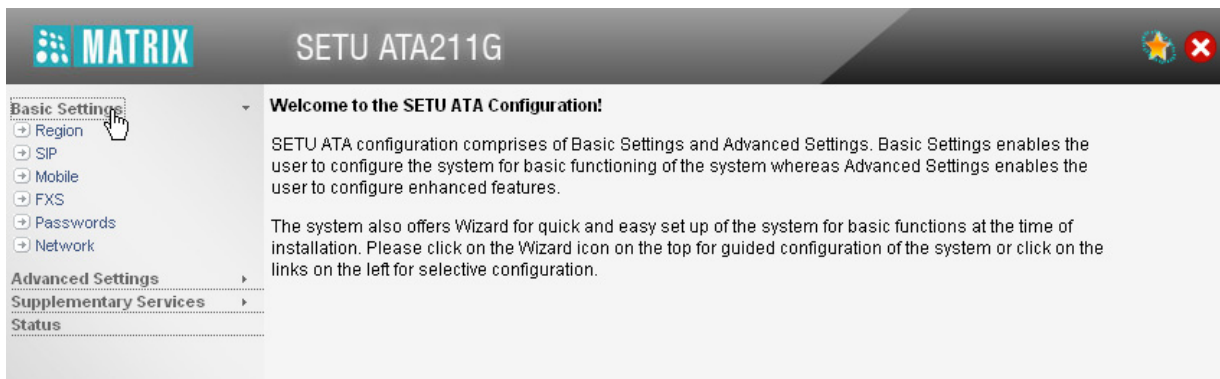
For a complete list of commands, refer “[System Commands](#)” in the Appendix.

The Basic Settings enable you to configure SETU ATA211G for basic functions. As Basic Settings cover much of your configuration requirements, you will be able to operate and use the system efficiently, when you configure Basic Settings.

To configure Basic Settings,

- Click **Basic Settings** link.

The links to the different basic parameters appear on the left pane.



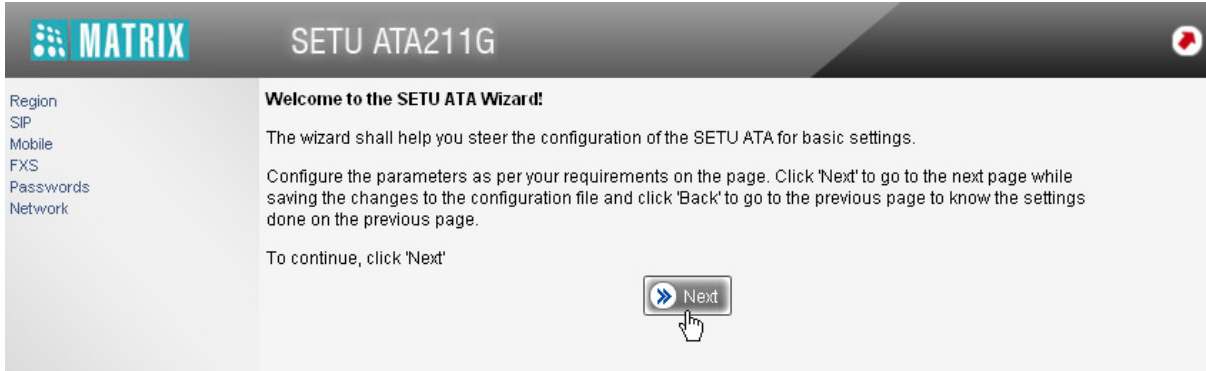
There are two ways to configure Basic Settings:

- Using the *Wizard*. The Wizard will guide you step-by-step through the configuration.

OR

- Selectively configuring the Basic Settings parameters, by clicking the links on the left pane.

To use the *Wizard*,



- Click the **Wizard**  icon on the top right of your screen.
- The **Next** button takes you to the next page, saving the changes you made on the current page.
- The **Back** button returns you to the previous page.
- The **More**  button expands parameters on the page.
- The **Less**  button collapses parameters on the page.
- The **Quit**  button allows you to exit the Wizard at any stage, saving changes you made before exiting.

To use selective configuration,

- Click **Basic Settings** link to expand.
- Click each parameter link, **Region**, **SIP**, **Mobile**, **FXS**, **Passwords**, **Network**.
- The selected parameter page opens.
- The **More**  button expands parameters on the page.
- The **Less**  button collapses parameters on the page
- Set the desired values on the page.
- Click **Submit** to save your settings on the page.

You may use the Wizard or selectively configure the Basic Settings pages, whichever works best for you.

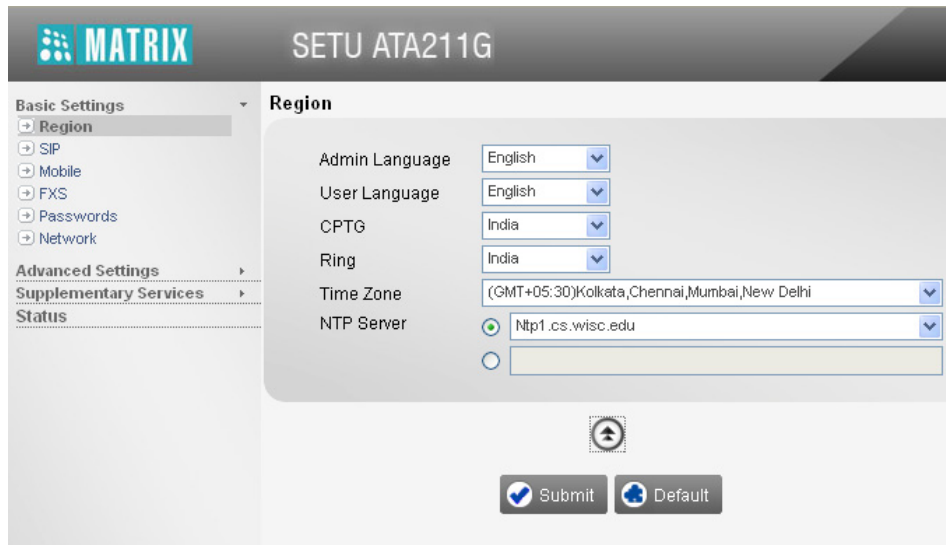
The instructions provided in this chapter, describe *selective* configuration of the Basic Settings pages.

1. Click **Basic Settings** to expand.
2. The following links appear under Basic Settings:
 - Region
 - SIP
 - Mobile
 - FXS
 - Passwords
 - Network

Each of these are explained in detail in the following.

Region

1. Click **Region**.



2. Click **More** button to view all parameters on the page.
3. Select **Admin Language** and **User Language**. Default: English.

SETU ATA211G can display the Admin and User pages of the GUI, Jeeves, in the following languages:

- English
- Italian
- Spanish
- French
- German
- Portuguese

All the Admin and User Pages of the GUI will appear in the language you selected, when you login again as Admin or User.

You can also select a Language on the Login page; however, it will be applied for the current session only.

4. Select **CPTG** (Call Progress Tone Generation) to match the CPTG of the country where SETU ATA211G is installed. Default: India.

The ATA211G supports country specific Call Progress Tones Generation (CPTG) to simulate the same tones of the local PSTN to which it is connected. The CPTG supported by SETU ATA211G for different countries are presented in the ["Appendix"](#).



When you reset SETU ATA211G to factory defaults, CPTG you selected will not be set to default.

5. Select **Ring** cadence to match the ring cadence of the country where SETU ATA211G is installed. Select your country from the drop down list. Default: India.

During an incoming call, SETU ATA211G will play the ring over the SLT as per the Ring Cadence of the country you selected. For a complete list of Ring Cadences of different countries supported by SETU ATA211G, see ["Appendix"](#).



When you reset SETU ATA211G to factory defaults, the Ring you selected will not be set to default.

6. To synchronize Date and Time of SETU ATA211G with that of the country where it is installed, select **Time Zone** from the drop down list. Default: (GMT+05:30) Kolkata, Chennai, Mumbai, New Delhi.
7. To use a public internet time server for date and time synchronization, select an internet based time server as **NTP Server**. You may select any of the three reliable public internet time servers¹ supported by SETU ATA211G:
 - ntp1.cs.wisc.edu
 - time.windows.com
 - time.nist.govDefault: Ntp1.cs.wisc.edu

If you want to use an NTP server other than these, select the radio button and enter the IP Address/ domain of the server in the empty field.



When you reset SETU ATA211G to factory defaults, Date and Time settings will not be set to default.

8. Click **Submit** to save your region settings.

1. You can select from three free, reliable public internet time servers run by the University of Wisconsin-Madison, Microsoft, and the National Institute of Standards and Technology (NIST), to obtain date and time. You can also configure an NTP time server of your preference, other than these. These public internet time servers provide time offset from the Greenwich Mean Time (GMT), and you can select the time according to the time zone of the country you are installing the SETU ATA211G. For instance, if your ATA211G is installed in India, you can select the time zone for India (GMT+5.30 Calcutta, Chennai, Mumbai, and New Delhi). The time for India is offset from GMT by +5.30 hours. Similarly, if your ATA211G is installed in Hawaii, select the time zone for Hawaii, which is offset by GMT -10:00 hours, to set the correct time and date.

SIP Trunks

SETU ATA211G supports three SIP trunks, which may be configured as Proxy or Non-Proxy, i.e. Peer-to-Peer trunks. You may register with three different ITSPs and use their services.

1. Click the **SIP** link.

The parameters of SIP Trunk 1 appear on your screen.

The screenshot displays the web interface for the SETU ATA211G device. The top header shows the 'MATRIX' logo and the device name 'SETU ATA211G'. On the left, a sidebar menu lists 'Basic Settings' (with sub-items: Region, SIP, More, FXS, Passwords, Network), 'Advanced Settings', 'Supplementary Services', and 'Status'. The 'SIP' option is highlighted. The main content area is titled 'SIP Trunk 1' and contains two sections: 'Trunk Settings' and 'Incoming Call'. The 'Trunk Settings' section includes fields for 'SIP Trunk' (with an 'Enable' checkbox), 'SIP Trunk Mode' (radio buttons for 'Proxy' and 'Peer-to-Peer', with 'Peer-to-Peer' selected), 'SIP ID', 'Authentication ID', 'Authentication Password', 'SIP Registration' (checked 'Enable'), 'Outgoing Calls Without Registration' (unchecked 'Allow'), 'Registrar Server Address:Port' (with a port of 5060), 'Outbound Proxy' (unchecked 'Enable'), 'Outbound Proxy Address:Port' (with a port of 5060), 'Instance in REGISTER?' (checked 'Add'), 'Re-Registration Timer' (3600 sec), and 'Registration Retry Timer' (10 sec). The 'Incoming Call' section shows 'Incoming Call Routing Options' with radio buttons for 'Reject Incoming Calls' and 'Route to FXS', with 'Route to FXS' selected.

2. Click **More** button to view all parameters of SIP Trunk 1 on this page.

Trunk Settings

Basic Settings

- Region
- SIP**
- Mobile
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Trunk Settings

SIP Trunk 1 SIP Trunk 2 SIP Trunk 3

SIP Trunk ☐ Enable

SIP Trunk Mode ☐ Proxy ☒ Peer-to-Peer [+](#)

SIP ID

Authentication ID

Authentication Password

SIP Registration ☒ Enable

Outgoing Calls Without Registration ☐ Allow

Registrar Server Address:Port : 5060

Outbound Proxy ☐ Enable

Outbound Proxy Address:Port : 5060

'instance' in REGISTER? ☒ Add

Re-Registration Timer 3600 sec

Registration Retry Timer 10 sec

1. Select the **SIP Trunk Enable** check box to use the SIP Trunk. Default: Disabled.

You may disable the SIP trunk, if you do not want to include it in Call Routing.

2. Select **SIP Trunk Mode** according to your installation. Default: Peer-to-Peer.
 - Select **Proxy**, if you want to register this SIP trunk with an ITSP or a Registrar Server.
 - Select **Peer-to-Peer**, if you want to use the trunk for Peer-to-Peer (non-proxy) calls.

If you select Peer-to-Peer, you must also configure the number strings in the Peer-to-Peer Table.

- Click the arrow icon.

Basic Settings

- Region
- SIP**
- Mobile
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Trunk Settings

SIP Trunk 1 SIP Trunk 2 SIP Trunk 3

SIP Trunk ☐ Enable

SIP Trunk Mode ☐ Proxy ☒ Peer-to-Peer [+](#)

SIP ID

Authentication ID

Authentication Password

SIP Registration ☒ Enable

Outgoing Calls Without Registration ☐ Allow

Registrar Server Address:Port : 5060

Outbound Proxy ☐ Enable

Outbound Proxy Address:Port : 5060

'instance' in REGISTER? ☒ Add

Re-Registration Timer 3600 sec

Registration Retry Timer 10 sec

Click to Configure Peer-to-Peer Numbers

- The **Peer-to-Peer Numbers** table opens in a new window.

Peer-to-Peer Numbers

Index	Number	Name	Minimum Digits	Maximum Digits	Destination Address
001	No Match Found		01	24	
002			01	24	
003			01	24	
004			01	24	
005			01	24	
006			01	24	
007			01	24	
008			01	24	
009			01	24	
010			01	24	
011			01	24	
012			01	24	
013			01	24	
014			01	24	
015			01	24	

Submit Default

- You can configure as many as 500 number strings, which are stored against an Index number.
- In the **Number** field, enter the peer-to-peer number string - prefix or entire number - that will be dialed. The number string must not exceed 24 characters. Default: Blank.

If the number to be dialed out is <dialednumber@destination address>, for example, 123@abc.com, you must enter 1234 in this field.

- To identify the number string you configured, enter a name in the **Name** field. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.
- As **Minimum Digits**, define the minimum length of the number string that must be dialed for the system to consider it as a valid number. Default: 01.

If the peer-to-peer number string you dial is shorter than the Minimum Digits you have configured, the system will not dial out the number.

- As **Maximum Digits**, define the maximum length of the number string that must be dialed out for the system to consider it the complete number string. Default: 24.

If the peer-to-peer number string you dial is longer than the Maximum Digits you have configured, the system will strip off the additional digits and dial out the number.

- In the **Destination Address** field, enter the domain name or IP Address to where the dialed peer-to-peer number string is to be sent. The Destination Address may consists of up to 40 characters. Default: Blank.

For example, if the peer-to-peer number to be dialed out is 123@abc.com, enter abc.com as Destination Address. If the number is 1234@ 192.168.1.197, enter 192.168.1.197 as the Destination Address. The Destination Address can also be in the form of Address:Port number.

- Click **Submit** to save entries.
- Close the window.

3. Enter the **SIP ID**. This is the ID which remote parties will use to call this SIP Trunk. Default: Blank.

The SIP ID may be a number or text consisting of a maximum of 40 characters.

If you have defined the trunk mode as *Proxy*, enter the SIP ID provided by your ITSP. For example, if SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field.

If you have defined the trunk mode as *Peer-to-Peer*, enter the desired SIP ID.

4. Enter **Authentication ID**. Default: Blank.

If you have defined the SIP trunk mode as *Proxy*, enter the Authentication ID (User ID) provided by your ITSP.

If you have defined the trunk mode as *Peer-to-Peer*, enter the ID of your preference as Authentication ID.

5. Enter **Authentication Password**. Default: Blank.

If you have defined the SIP trunk mode as *Proxy*, enter the Authentication Password provided by your ITSP.

If you have defined the trunk mode as *Peer-to-Peer*, enter a password of your preference as Authentication Password.

6. **SIP Registration** is applicable for Proxy SIP Trunks only. Default: Disabled.

Select the check box to enable. SETU ATA211G will send the REGISTER message to Registrar proxy or Outbound proxy as applicable.

7. Select the **OG Calls without Registration** check box to allow the users to make outgoing calls irrespective of whether the SIP Trunk has been successfully registered with proxy or not.

By default, the system does not allow outgoing calls to be made if the status of the SIP trunk is 'not registered'.

8. If you have defined the SIP trunk mode as Proxy, in the **Registrar Server Address: Port** field, enter the Registrar Server Address and the Registrar Server's listening port for SIP messages.

The registrar server address may be an IP address or a domain. Default: Blank.

The valid port range is: 1024-65534. Default: 5060.

9. If your service provider uses an outbound proxy for handling voice calls, enable the **Outbound Proxy** check box.

- In the **Outbound Proxy Server Address: Port** field, enter the IP address or domain name of the Outbound Proxy Server and the number of the Outbound Proxy Server's Listening Port for SIP.

The Server Address may consist of maximum 40 characters. Default: Blank.

The valid range for the port is 1024-65534. Default: 5060.

10. To add '**rinstance**' in the REGISTER Message, keep this feature enabled. Default: Disabled.

'rinstance' is any random value which can be used by the SETU ATA211G to fetch its own contact binding, i.e. to know the Registration Expiry Timer assigned by the server.

When you enable 'rinstance' in Register, SETU ATA211G will generate any random value of 'rinstance' and include in the REGISTER message. The system will use the registration expiry timer of that contact binding.

11. Set the **Re-registration Timer**. This is the time period after which the SETU ATA211G will send registration request to maintain registration binding with the Registrar server.

The valid range of this timer is 00001- 65535. Default: 3600 seconds.

12. Define the **Registration Retry Timer**. When a registration attempt fails, SETU ATA211G will resend registration request to the Registrar Server after the expiry of the Re-registration Timer. The valid range of this timer is from 00001- 65535. Default: 10 seconds.

13. Click **Submit**.

Incoming Call

Basic Settings

- Region
- SIP**
- Mobile
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Incoming Call

Incoming Call Routing Options

- ☐ Reject Incoming Calls
- ☒ Route to FXS
- ☐ Route to fixed destination number through Mobile
- ☐ Route to FXS. If not answered within ring timer, route to fixed destination number through Mobile
- ☐ Answer the call, collect number & route to FXS
- ☐ Answer the call, collect number & route to Mobile
- ☐ Route as per Called Party Number

Fixed Destination Number

Ring Timer sec

CLI number on FXS port ☒ Calling Number ☐ Called Number

On Connecting Media, Send ☒ 183 Session Progress ☐ 200 OK

Authenticate Incoming Calls ☒ Disable ☐ PIN ☐ Digest

1. Select an appropriate Incoming Call Routing option for the SIP Trunk. You may select:

Reject Incoming Calls: The system will reject all calls received on the SIP Trunk. Select this option, if you want to block calls on this SIP trunk.

- **Route To FXS:** The system will place all calls received on the SIP Trunk on the FXS port. The phone connected to the port will start ringing for the duration of the Ring Timer (default: 45 seconds). If the call is not answered within this timer, it will be rejected. This is also the default routing option.
- **Route to fixed destination number through Mobile:** The system will place all calls received on the SIP Trunk to a Fixed Destination Number through the Mobile port.

If you select this option, define the destination number in the **Fixed Destination Number** field.

- **Route to FXS. If not answered within ring timer, route to fixed destination number through Mobile:** The system will place all calls received on the SIP Trunk on the FXS port. The phone connected to the port will start ringing for the duration of the **Ring Timer** (default: 45 seconds). If the call is not answered within this timer, the call will be automatically routed to the Fixed Destination Number through the Mobile port.

If you select this option, define the destination number in the **Fixed Destination Number** field.

- **Answer the call, collect number & route to FXS:** Select this option, if FXS port of SETU ATA211G is connected to the FXO port of the PBX and you want callers to reach extensions of the PBX by simply dialing the extension number. When there is an incoming call on the SIP trunk, the call is answered and dial tone is played to the caller. When the caller dials the number, ATA will collect the digits and route the call from its FXS port.
- **Answer the call, collect number & route to Mobile:** Select this option, if you want the system to first answer the call on the SIP trunk, play dial tone to the caller, and when the caller dials the number, to collect the digits and route the call to the Mobile port.
- **Route as per called party number:** Select this option, if you want all incoming calls on the SIP trunk to be routed on the basis of the Called Party Number received in INVITE message.

If you select this option, you must configure the **Called Party Number Table**. See [“Called Party Number Table”](#) for instructions.

- Enter the **Fixed Destination Number** in this field, if the incoming call routing option requires the call to be finally placed at a fixed destination. You can configure the fixed destination number of maximum 16 digits. Default: Blank
 - Enter the time duration in the **Ring Timer** field, for which the system plays the ring on the FXS Port for an incoming call. Range of the ring timer is of 01 to 99 seconds. Default: 45 sec.
2. SETU ATA211G supports display of Calling Line Identity (CLI) received in the SIP request message on the FXS port. To display **CLI Number on FXS Port**, you may select:
 - Calling Number
 - Called Number
 Default: Calling Number
 3. Define the type of response the system should send on the receipt of an incoming call on the SIP trunk. For the **On Connecting Media, Send** parameter, you may select:
 - 183 Session Progress
 - 200 OK
 Default: 183 Session Progress

4. If you want to **Authenticate Incoming Calls** on SIP, that is, you want to allow incoming calls only after callers have authenticated themselves. You can select:

- **PIN:** Select this option to authenticate callers with a unique PIN.
- **Digest:** Select this type of authentication if you want to allow incoming calls on the SIP Trunk only after callers have authenticated themselves with their User ID and Password.
Default: Disabled.

If you have selected **PIN**, you must configure the PIN Authentication Table. To do this,

- Click the arrow icon.

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP**
- Mobile
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Incoming Call

Incoming Call Routing Options

- ☐ Reject Incoming Calls
- ☒ Route to FXS
- ☐ Route to fixed destination number through Mobile
- ☐ Route to FXS. If not answered within ring timer, route to fixed destination number through Mobile
- ☐ Answer the call, collect number & route to FXS
- ☐ Answer the call, collect number & route to Mobile
- ☐ Route as per Called Party Number

Fixed Destination Number

Ring Timer sec

CLI number on FXS port ☒ Calling Number ☐ Called Number

On Connecting Media, Send ☒ 183 Session Progress ☐ 200 OK

Authenticate Incoming Calls ☐ Disable ☒ PIN ☐ Digest

Vocoders

Click to Configure PIN Authentication

- The **PIN Authentication** page opens in a new window.

PIN Authentication

Index	PIN	Index	PIN
001		051	
002		052	
003		053	
004		054	
005		055	
006		056	
007		057	
008		058	
009		059	
010		060	
011		061	
012		062	
013		063	
014		064	
015		065	
016		066	
017		067	
018		068	
019		069	
020		070	

- Enter the PIN to be authenticated against each Index in the Table.

The PIN number can have a maximum of 4 digits. The characters allowed are: 0-9, *, #

- If you have finished entering, click **Submit** to save.
- Close the window. See [“PIN Authentication”](#).

If you enable **Digest Authentication**, you must configure the **Digest Authentication Table**. To do this,

- Click the arrow icon.

- The **Digest Authentication Table** page opens in a new window.

Index	User ID	User Password
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		
013		
014		
015		
016		
017		
018		

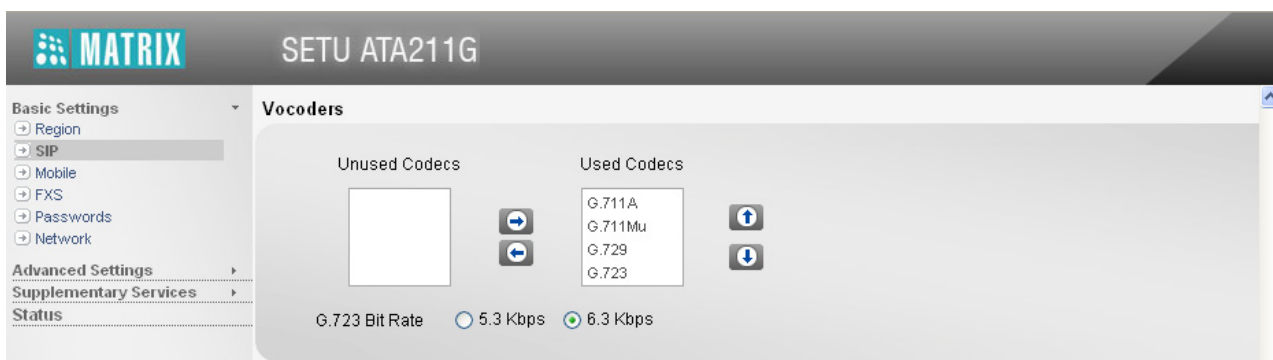
- Enter the **User ID** to be authenticated along with its corresponding **Password** against each Index in the Table.

The User ID may be a maximum of 40 characters. The User Password may be a maximum of 16 characters.

- If you have finished entering the User IDs and their corresponding User Passwords in this Table, click **Submit**.
- Close the window.

To know more about this feature, see [“Digest Authentication”](#).

Vocoders



1. Select **Vcoders** in the order of preference from the multiple selection box.

Vcoders are the various voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality. You can set 4 Vocoder options in the order of preference.

The Vcoders supported by SETU ATA211G in the order of preference, i.e. 1st to 4th, listed in the **Used Codecs** box are:

- G.711 A - Law
 - G.711 μ - Law
 - G.729
 - G.723
- To remove a Vocoder from this list, select the Vocoder and click the back arrow $\leftarrow$
 - The Vocoder will be moved to the **Unused Codecs** box.
 - Select the required Vcoders your cursor from the right list box.
 - Use the **Up** and **Down** arrows near the **Used Codecs** box to change the order of Vocoder preference.
2. If you have G.723 as a Preferred Vocoder, select **G.723 Bit Rate** as **5.3 Kbps** or **6.3 Kbps**. Default: 6.3kbps.

When G.723 is negotiated, the selected Bit Rate will be applied only when sending the RTP packets. When receiving RTP packets from the remote end, both Bit Rates of G.723 will be accepted.

Advanced

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP**
- Mobile
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Advance

SIP Transport: ☒ UDP ☐ TCP ☐ TCP (Fallback to UDP)

Automatic Number Translation: ☒ Enable

CLIR: ☒ Enable

CLI Source: ☒ FXS/Mobile

Anonymous Call Rejection: ☒ Enable

Symmetric RTP: ☒ Enable

DNS SRV: ☒ Enable

NAT Type: ☒ Disable ☐ STUN ☐ Router IP Address

DTMF: ☐ Inband ☒ Outband ☐ SIP INFO

On Time: 100 msec

Off Time: 100 msec

Pause Timer: 2 sec

Call Hold Using: ☒ RFC 3261 ☐ RFC 2543

FAX Protocol: ☒ T.38 (UDPTL) ☐ Pass Through

Returned Calls to Original Caller (RCOC)

On Busy: ☐ Enable

On No Response: ☐ Enable

On Speech: ☐ Enable

Submit Default

1. Select the default **SIP Transport** for outgoing SIP messages from the following options:

- **UDP:** Outgoing messages are transported using UDP.
- **TCP:** Outgoing messages are transported using TCP.
- **TCP (Fallback to UDP):** TCP is used for outgoing messages. However, if the TCP connection fails, the system will attempt to send the message again over UDP.

Default: UDP



To use **TCP** or **TCP (Fallback to UDP)**, you must enable **SIP over TCP** in the [“System Parameters”](#) page.

2. Select the **Automatic Number Translation** check box, if you want to apply this feature on the SIP trunk.
Default: Disabled.

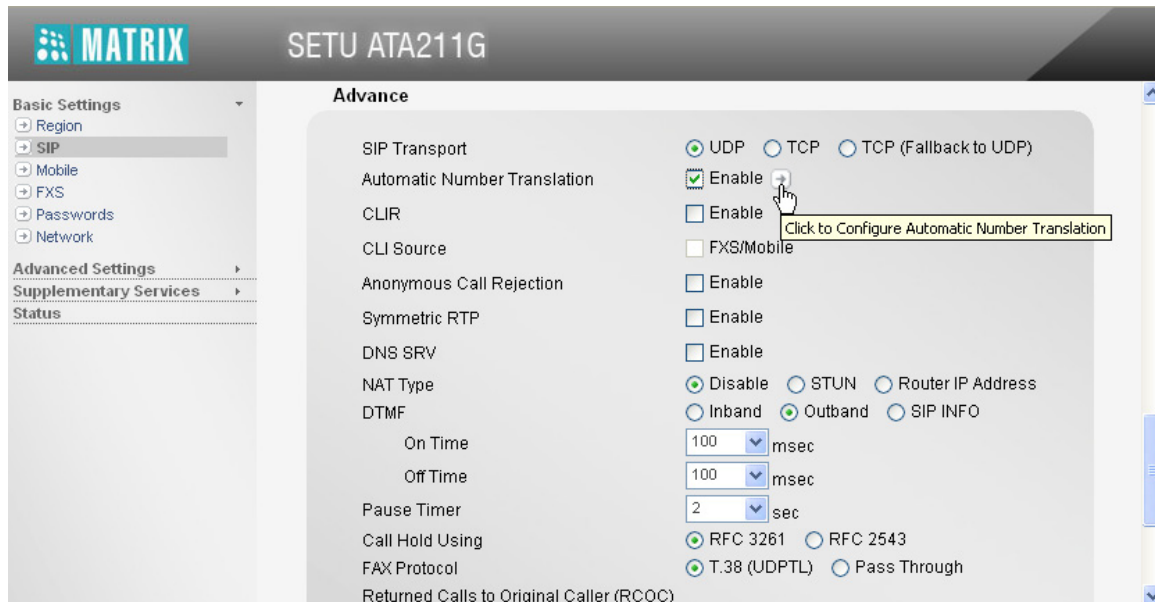
Automatic Number Translation (ANT) of SETU ATA211G modifies the number strings - entire numbers or part thereof - you dial into number strings understood by the IP network.

The ANT feature is useful when the network requires adding or stripping off of parts of the dialed number strings. For example, the PSTN requires you to dial the prefix 00 for calling international numbers, whereas the ITSP you have subscribed this SIP trunk restricts the dialing of the prefix 00 for international numbers. If you happen to dial this prefix, your call will be rejected by the ITSP.

ANT is also used when SETU ATA211G is deployed for networking between PBXs at multiple sites, and “Multi Stage Dialing” is required.

To use ANT, select the check box and configure the Automatic Number Translation Table. To do this,

- Make a list of numbers that need to be modified before being dialed out from this SIP trunk. Make a list of the corresponding modified numbers that the system should substitute the dialed numbers with.
- Click the arrow icon.



- The **Automatic Number Translation** table opens in a new window. In this table, you can store as many as 24 Dialed Numbers and their corresponding Substitute Numbers, at Index Numbers 01 to 24.

Automatic Number Translation		
Index	Dialed Number	Substitute Number
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		
20		
21		
22		
23		
24		

Submit Default

- In the **Dialed Number** column, enter the numbers that need to be modified when dialed out from this SIP trunk.
 - In the **Substitute Number** column, enter the numbers which the system should dial out in place of the dialed numbers.
Make sure you enter a Dialed Number and its corresponding Substitute Number at the same Index number in the table. For example, if you entered 001 as Dialed Number at Index 01, you must enter its corresponding Substitute Number 1 also at Index 01.
 - Click **Submit** to save your entries.
 - Close the window.
3. By default, the CLI of the SIP Trunk is sent to the called party when outgoing calls are made using the SIP trunk. If you do not want to send CLI, enable **CLIR** check box.
 4. Select the **FXS/Mobile** check box to select the CLI source type you want ATA to send (in the SIP message) during an outgoing call as the CLI.

If FXS is the source port, the FXS Port Number is sent as CLI. If Mobile is the source port, the same CLI received on the port is sent.
 5. If you want the system to reject incoming calls which do not have the identity of the caller (Calling Party Information), enable the **Anonymous Call Rejection** check box. Default: Disabled.
 6. If you want the system to send RTP packets to original IP and Port from where RTP packets are received, by ignoring the contact information received in SDP, enable the **Symmetric RTP** check box. Default: Disabled.
 7. If you want the system to send DNS SRV query to the configured domain server, enable **DNS SRV**. When disabled, the system will send DNS A query to the configured domain server. Default: Disabled.
 8. When the system is installed behind a NAT Router, select specific NAT traversal mechanism to be used as **NAT Type**.
 - Select **Router's IP Address**, if your ATA is located behind the NAT router (any type).

Make sure you disable Outbound Proxy on SIP trunk and configure the same IP Address under NAT settings in the "[System Parameters](#)" page.
 - Select **STUN** if your ATA is located behind the NAT router other than Symmetric.

Make sure you disable Outbound Proxy on SIP trunk and configure the STUN Server Address and port in "[System Parameters](#)".
 9. Select the appropriate **DTMF** sending/receiving mechanism that is compatible with the DTMF sending/receiving mechanism of your ITSP or remote peer. SETU ATA211G supports:
 - Inband
 - Outband
 - SIP (INFO)
 Default: Outband

- In the **On Time** field, set the time for which the DTMF digit should remain on, when the digit is dialed out by the system. The DTMF On Time may be 100 or 150 or 200 milliseconds. Default: 100 milliseconds.
- In the **Off Time** field, set the time for which system should wait before dialing the successive DTMF digits. The DTMF Off Time may be 100 or 150 or 200 milliseconds. Default: 100 milliseconds.

10. Configure the **Pause Timer (sec)**, if you want to insert a delay before the digits of a number string is out dialed from the SIP trunk.

The Pause Timer is typically used in [“Multi Stage Dialing”](#). The range of the Pause Timer is from 1 to 9 seconds. Default: 2 seconds.

11. Select an appropriate **Call Hold Method** that is compatible with your ITSP proxy server/remote peer. You may select:
 - RFC 3261
 - RFC 2543
 Default: RFC 3261

12. To send and receive the Fax over IP from and on the FAX machine connected to FXS port of the SETU ATA211G, select the desired Fax over IP protocol:

- **T.38 (UDPTL):** If you select this option, the device you are sending the fax to, must also support this protocol.
- **Pass Through:** Select this option if you need to send fax over G.711. The device you are sending fax to must also use G.711.

Default: T.38 (UDPTL).



If the FAX sent using T.38 is rejected, SETU ATA211G will use Pass Through as the Fall Back protocol to send the fax.

13. In the **Returned Calls to Original Caller (RCOC)** option, enable the type of RCOC you want to set for the SIP Trunk. See [“Return Call to Original Caller \(RCOC\)”](#)
 - **On Busy:** Enable this type if you want to set RCOC when the call is returned if the called party is found busy.
 - **On No Response:** Enable this type if you want to set RCOC when the call is returned if the called party does not answer the call.
 - **On Speech:** Enable this type if you want to set RCOC when the call is returned after speech was established with the called party.
14. If you have completed the configuration of SIP Trunk 1, click **Submit** to save settings.
15. To configure another SIP trunk, click the **SIP Trunk Number** tab.
16. Follow the same instructions as described here to configure the next SIP trunk.

Mobile Port

SETU ATA211G supports a Mobile Port. Insert the SIM card in the SIM holder to get your ATA211G connected to the GSM network.

To configure the Mobile Port parameters,

1. Click **Mobile**.

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

General

Mobile Port ☐ Enable

Name

Incoming Call

Incoming Call Routing Options

- ☐ Reject Incoming Calls
- ☒ Route to FXS
- ☐ Route to fixed destination number as per incoming call route selection
- ☐ Route to FXS. If not answered within ring timer, route to fixed destination number as per incoming call route selection
- ☐ Answer the call, collect number & route as per incoming call route selection
- ☐ Route to FXS. If not answered within ring timer, answer the call, collect number & route as per incoming call route selection

Fixed Destination Number

Incoming call route ☒ SIP1 ☐ SIP2 ☐ SIP3 ☐ Dialed Number Table

Ring Timer sec

CLI Number on FXS port ☒ Calling Number ☐ Called Number

Authenticate Incoming Calls ☐ PIN Authentication

Advance

Automatic Number Translation ☐ Enable

2. Click **More** button to view all Mobile parameters on this page.

General

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

General

Mobile Port ☐ Enable

Name

Incoming Call

Incoming Call Routing Options

- ☒ Reject Incoming Calls

1. Select the **Enable** check box to use the Mobile Port. Default: Disabled.

You may disable the Mobile Port, if you do not want to include it in Call Routing.

2. You can assign a **Name** to the Mobile port, which will be displayed to the called party (if the instrument of the called party supports display name functionality).

The name you assign may consist of a maximum of 24 characters. Default: Blank.

Incoming Call

1. Select an appropriate Incoming Call Routing option for the Mobile Port. You may select:

- **Reject Incoming Calls:** Select this option if you want ATA211G to reject the incoming calls on the Mobile port.
- **Route to FXS:** Select this option if you want to route the incoming calls on Mobile port, to the FXS port. The incoming call will be placed on the FXS Port and the phone connected to the FXS Port will start ringing for the duration of the **Ring Timer** (default: 45 seconds). If the call is not answered within this timer, it will be rejected. This is also the default option.

If you select this option, you may configure the **Ring Timer**. Valid range of this timer is 01-99 seconds.

- **Route to fixed destination number as per incoming call route selection:** Select this option if you want incoming calls on the Mobile Port to be routed to a Fixed Destination Number as per the Incoming Call Route you select: SIP1, SIP2, SIP3 or Dialed Number Table.

If you select this option, define the destination number in the **Fixed Destination Number** field and select the **Incoming Call Route**.

- **Route to FXS. If not answered within ring timer, route to fixed destination number as per incoming call route selection:** Select this option if you want to route the incoming calls on the Mobile Port to the FXS Port first. The phone connected to the FXS Port will start ringing for the duration of the **Ring Timer** (default: 45 seconds). If the call is not answered within this timer, the call will be automatically routed to the Fixed Destination Number as per the Incoming Call Route you select: SIP1, SIP2, SIP3 or Dialed Number Table.

If you select this option, define the destination number in the **Fixed Destination Number** field and select the **Incoming Call Route**.

- **Answer the call, collect number & route as per incoming call route selection:** Select this option if you want the system to first answer the call on the Mobile Port, play dial tone to the caller, collect the digits dialed by the caller and route the call as per the Incoming Call Route you select: SIP1, SIP2, SIP3 or Dialed Number Table.

If you select this option, select the **Incoming Call Route**.

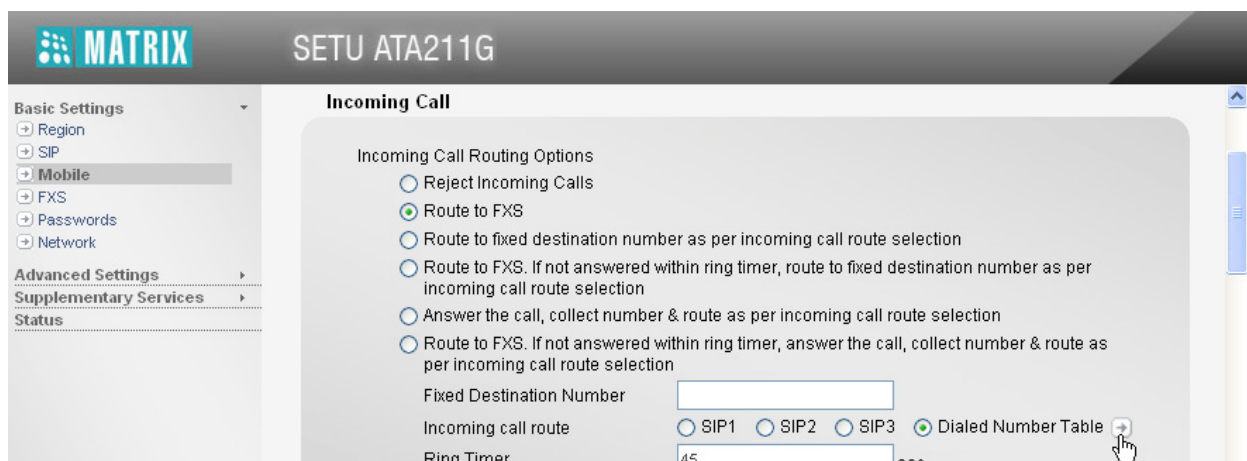
- **Route to FXS. If not answered within ring timer, answer the call, collect number & route as per incoming call route selection:** Select this option if you want to route the incoming calls on the Mobile Port to the FXS Port first. The phone connected to the FXS Port will start ringing for the duration of the Ring Timer (default: 45 seconds). If the call is not answered within this timer, the system answers the call, plays dial tone to the caller. When the caller dials the number, the system collects the digits and routes the call as per the Incoming Call Route you select: SIP1, SIP2, SIP3 or Dialed Number Table.

If you select this option, select the **Incoming Call Route**. You may also configure the **Ring Timer**.

- **Fixed Destination Number:** Enter the fixed destination number, if the incoming call routing option you selected requires the call to be finally placed to a fixed destination number.

This destination number can be configured of maximum 40 characters.

- **Incoming Call Route:** You need to configure this parameter, only if the Incoming Call Routing option you selected above requires the **Incoming Call Route** to be selected. You may select:
 - **SIP1/SIP2/SIP3:** Select this option as Incoming Call Route if you want to route the incoming call received on Mobile port using SIP1/SIP2/SIP3 trunk.
 - **Dialed Number Table:** Select this option Incoming Call Route if you want SETU ATA211G to route the incoming call on the basis of the number dialed. If you select this option, you must also configure the Dialed Number Table. To do this,
 - Click the arrow icon.



- The **Dialed Number Table** opens in a new window.

001-025
026-050
051-075
076-100

Dialed Number Table

Index	Number	Minimum Digits	Maximum Digits	Destination Route
001	0	01	24	SIP1
002	1	01	24	SIP1
003	2	01	24	SIP1
004	3	01	24	SIP1
005	4	01	24	SIP1
006	5	01	24	SIP1
007	6	01	24	SIP1
008	7	01	24	SIP1
009	8	01	24	SIP1
010	9	01	24	SIP1
011		01	24	SIP1
012		01	24	SIP1
013		01	24	SIP1
014		01	24	SIP1
015		01	24	SIP1
016		01	24	SIP1
017		01	24	SIP1

If dialed number does not match with either of the programmed number in dialed number table & call is made from FXS, route the call through

☐ None
 ☒ SIP1
 ☐ SIP2
 ☐ SIP3
 ☐ Mobile

If dialed number does not match with either of the programmed number in dialed number table & call is made from Mobile, route the call through

☐ None
 ☒ SIP1
 ☐ SIP2
 ☐ SIP3
 ☐ FXS

☒ Submit

☐ Default

For each Index number,

- Enter the desired **Number** you want to store in the table. The number must not exceed 24 characters.
- For the number you entered, define the **Minimum Digits**, which the system should wait to receive before considering it as a valid number. Default: 01.
- For the number you entered, define the **Maximum Digits**, which the system should wait to receive before considering it as End of Dialing. Default: 24.
- For the number you entered, select as **Destination Route** the port the system should use to call the dialed number. Default: SIP 1.

At the bottom of the table, select the destination route for dialed numbers that do not match with any of the entries on this table.

- For calls made from the FXS port, select the the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from FXS, route the call through**. Default, SIP Trunk 1.
- For calls made from the FXS port, select the the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from Mobile, route the call through**. Default, SIP Trunk 1.

- Click **Submit** to save entries and close the window. See [“Dialed Number Table”](#).
2. In the **Ring Timer** field, set the duration for which the ring should be played on the FXS Port for an incoming call. The range of the Ring Timer is of 01 to 99 seconds. Default: 45 sec.
 3. Select the required type of **CLI to be displayed on the FXS Port** during an incoming call:
 - **Calling Number:** Select this option if you want the number of the Calling Party to be displayed as CLI on the FXS Port.
 - **Called Number:** Select this option if you want the number of the Called Party to be displayed as CLI on the FXS Port.
 4. If you want to **Authenticate Incoming Calls** on the Mobile port only after callers have authenticated themselves, enable the **PIN Authentication** check box. Default: Disabled.

You must configure the PIN Authentication Table. To do this,

- Click the arrow icon.

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Incoming Call

Incoming Call Routing Options

- ☐ Reject Incoming Calls
- ☒ Route to FXS
- ☐ Route to fixed destination number as per incoming call route selection
- ☐ Route to FXS. If not answered within ring timer, route to fixed destination number as per incoming call route selection
- ☐ Answer the call, collect number & route as per incoming call route selection
- ☐ Route to FXS. If not answered within ring timer, answer the call, collect number & route as per incoming call route selection

Fixed Destination Number

Incoming call route ☐ SIP1 ☐ SIP2 ☐ SIP3 ☒ Dialed Number Table

Ring Timer sec

CLI Number on FXS port ☒ Calling Number ☐ Called Number

Authenticate Incoming Calls ☒ PIN Authentication

- The **PIN Authentication** table opens in a new page.

PIN Authentication

Index	PIH	Index	PIH
001		051	
002		052	
003		053	
004		054	
005		055	
006		056	
007		057	
008		058	
009		059	
010		060	
011		061	
012		062	
013		063	
014		064	
015		065	
016		066	
017		067	
018		068	
019		069	
020		070	

- Enter the PIN to be authenticated against each Index in the Table.

The PIN number can have a maximum of 4 digits. The characters allowed are: 0-9,*, #

- If you have finished entering, click **Submit** to save.
- Close the window. See [“PIN Authentication”](#).

Advanced

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- **Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Advance

Automatic Number Translation ☐ Enable

CLIR ☐ Enable

Mobile Network Selection ☒ Automatic ☐ Manual

Priority1

Priority2

Priority3

Priority4

Priority5

Priority6

Priority7

Priority8

Priority9

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Mobile Settings:

Priority9: []

Band: 900+1800 mhz

Transmit Gain: Normal

Receive Gain: Normal

SIM PIN: 1234

DTMF ON Time: 100 msec

DTMF OFF Time: 100 msec

DTMF Level: -6.0

Pause Timer: 1 sec

Returned Calls to Original Caller (RCOC):

- On Busy: ☐ Enable
- On No Response: ☐ Enable
- On Speech: ☐ Enable
- Replace '*' from the CLI: ☐ Yes
- Number String: []

Buttons: Submit, Default

1. Select the **Automatic Number Translation** check box, if you want to apply this feature on the Mobile port. Default: Disabled.

Automatic Number Translation (ANT) enables you to modify or translate the dialed number into another number. ANT is useful when you need to modify or replace a few digits of the dialed number with other digits, before the system dials out the number.

To apply ANT on the Mobile port, you must enable ANT and configure the Automatic Number Translation Table with dialed numbers and substitute numbers. To do this,

- Make a list of numbers that need to be modified before being dialed out from the Mobile port. Make a list of the corresponding modified numbers that the system should substitute the dialed numbers with.
- Click the arrow icon.

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile**
- FXS
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Advance Settings:

Automatic Number Translation: ☒ Enable

CLIR: ☐ Enable

Mobile Network Selection: ☒ Automatic ☐ Manual

Priority1: []

Priority2: []

Priority3: []

Priority4: []

Priority5: []

Tooltip: Click to Configure Automatic Number Translation

- The **Automatic Number Translation** table opens in a new window.

Index	Dialed Number	Substitute Number
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		
11		
12		
13		
14		
15		
16		
17		

Submit Default

In this table, you can store as many as 24 Dialed Numbers and their corresponding Substitute Numbers, at Index Numbers 01 to 24.

- In the **Dialed Number** column, enter the numbers that need to be modified when dialed out from the Mobile Port.
- In the **Substitute Number** column, enter the numbers which the system should dial out in place of the dialed numbers.

Make sure you enter a Dialed Number and its corresponding Substitute Number at the same Index number in the table. For example, if you entered 001 as Dialed Number at Index 01, you must enter its corresponding Substitute Number 1 also at Index 01.

- Click **Submit** to save your entries.
 - Close the window.
2. Enable **CLIR** check box to hide your identity from the called party. Thus, the outgoing calls made by the user on Mobile port will be sent as anonymous calls. CLIR will work only if it is supported by your service provider.
 3. In **Mobile Network Selection** option, set the network selection mode as **Automatic** or **Manual**.

When SETU ATA211G is located in the home region, the SIM card gets registered with the Network Operator of that SIM only, leading to no extra charges (roaming charges) to the user. Whereas, if SETU ATA211G is located in an area away from the home region or in an area where more than one Network Operators of different regions are available, the SIM card may get registered with another available Network resulting in roaming charges.

The Mobile Network Selection feature of SETU ATA211G enables you to select the GSM network of your choice, either **automatically** or **manually**. At Power On, ATA211G will automatically locate and register the SIM with the Network that supports the SIM card installed in it. If the SIM fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer.

- **Automatic:** Select this option if you want the system to select the GSM network automatically. This will register the SIM with any available GSM network whose signal strength is greater than that of other available networks.
- **Manual:** Select this option if you want to select the GSM network manually. This will allow you to get your SIM registered with the desired network operator available in that region.

If you select the Mobile Network Selection as **Manual**, assign priorities to the preferred network operators for the registration of the Mobile port in the fields **Priority 1** to **Priority 9**.

Enter Network Operator Codes (MCC and MNC) in the fields **Priority 1** to **Priority 9**.

In these fields, Network Operator Codes (**MCC and MNC**) are configured according to their priority. So, at every Power on, ATA211G will send a request for available GSM Network Operators, who in turn will respond to the request with a list of available Network Operator Codes. ATA211G will match the available Network Operator Codes with those configured in the fields **Priority 1** to **Priority 9** and select the Network Operator which best suits your requirement in order of priority. See "[Mobile Network Selection](#)".

4. In the **Band** field, select the GSM frequency band supported by your service provider in your country. ATA211G supports four major GSM frequency bands: **GSM 850, EGSM 900, DCS 1800** and **PCS 1900 mhz**. Default: 900 + 1800 mhz.

If band selection is changed, it will be effective only after ATA211G restarts.

5. In **Transmit Gain** field, set the Transmitting Gain of the RF Signal to optimum level by selecting one out of the five different levels: **Very Low, Low, Normal, High** and **Very High**. Default: Normal.
6. In **Receive Gain** field, set the Receiving Gain of the RF Signal to optimum level by selecting one out of the five different levels: **Very Low, Low, Normal, High** and **Very High**. Default: Normal.
7. In **SIM PIN** field, enter the unique SIM PIN for the Mobile port of your ATA211G. The SIM PIN you enter here must be of 4 minimum and 8 digits maximum. Default: 1234.

SIM PIN is a security feature used by the GSM network. If this feature is enabled on the SIM Card, the network asks the system for the SIM PIN at every power ON. If the SIM PIN you have programmed here matches with the one stored in the network, the network will mark ATA as present on the network.

8. In **DTMF ON Time** field, select the appropriate DTMF On time for sending DTMF digits. The DTMF On Time is the time for which the DTMF digit which is to be out dialed remains ON. Valid range: 50 to 200 milliseconds. Default: 100 milliseconds.
9. In **DTMF OFF Time** field, select the appropriate DTMF OFF time as the time gap between two DTMF digits. This can be set between 20 to 200 milliseconds. Default: 100 milliseconds.
10. In **DTMF Level** field, DTMF Level signifies with what magnitude (level/amplitude) the DTMF digits should be out dialed. This can be set between 0 dB to -10.5 dB. Default: -6 dB.

11. You may configure the **Pause Timer** if you want to insert a delay before dialing the DTMF digits on the Mobile port. The Pause Timer is typically used in Multi-Stage Dialing. Valid range: 1 to 9 seconds. Default: 2 seconds.
12. In the **Returned Calls to Original Caller (RCOC)** option, enable the type of RCOC you want to set for the Mobile Port.
 - **On Busy:** Enable this type if you want to set RCOC when the call is returned if the called party is found busy.
 - **On No Response:** Enable this type if you want to set RCOC when the call is returned if the called party does not answer the call.
 - **On Speech:** Enable this type if you want to set RCOC when the call is returned after speech was established with the called party.
13. Select **Replace '+' from CLI Yes** check box if you want to replace '+' from the CLI (during incoming call on mobile) with some other number string.

When an incoming call is received on the Mobile Port, the received CLI usually has a prefix "+". To replace this prefix with a number string, enable this check box. Default: Disabled.

To replace the prefix with another number you may configure **Number String**. You can configure the number string of maximum 6 digits.

If you want to remove '+', keep the Number String blank.

14. Click **Submit** to save the configurations.

FXS Port

SETU ATA211G supports an FXS port. Depending upon your requirement, you can either connect an standard, analog telephone instrument to the FXS port, or interface the FXS port with the FXO port of a PBX.

To configure the FXS port parameters,

1. Click **FXS**.

The screenshot shows the 'SETU ATA211G' web interface. On the left, the 'Basic Settings' menu is expanded, and 'FXS' is selected. The main content area is divided into sections: 'General', 'Outgoing Call', 'Class of Service', and 'Advance'. In the 'General' section, the 'FXS Port' is checked and labeled 'Enable'. Below it are input fields for 'Name' and 'Number'. The 'Outgoing Call' section has 'Outgoing Call Routing Options' with radio buttons for 'Don't allow outgoing calls', 'Route all calls from SIP2', and 'Route all calls from Mobile'. There are also checkboxes for 'Authenticate Outgoing Calls' (PIN Authentication) and 'Auto Mobile Fallback' (Enable). The 'Class of Service' section has a grid of checkboxes for 'Call Hold', 'Call Waiting', 'Attendant Transfer', 'Conference', 'Call Forward', 'Call Toggle', 'Blind Transfer', and 'Do-Not-Disturb', all of which are checked. The 'Advance' section is partially visible at the bottom.

2. Click **More** button to view all FXS port parameters on this page.

General

This screenshot shows the 'General' configuration page for the FXS port. The left sidebar is the same as the previous image. The main content area shows the 'General' section with 'FXS Port' checked and labeled 'Enable'. Below it are input fields for 'Name' and 'Number'. The 'Outgoing Call' section is visible below, showing 'Outgoing Call Routing Options'.

1. By default Call Routing is enabled on the FXS port.

You may disable the FXS Port and if you do not want to include it in Call Routing.

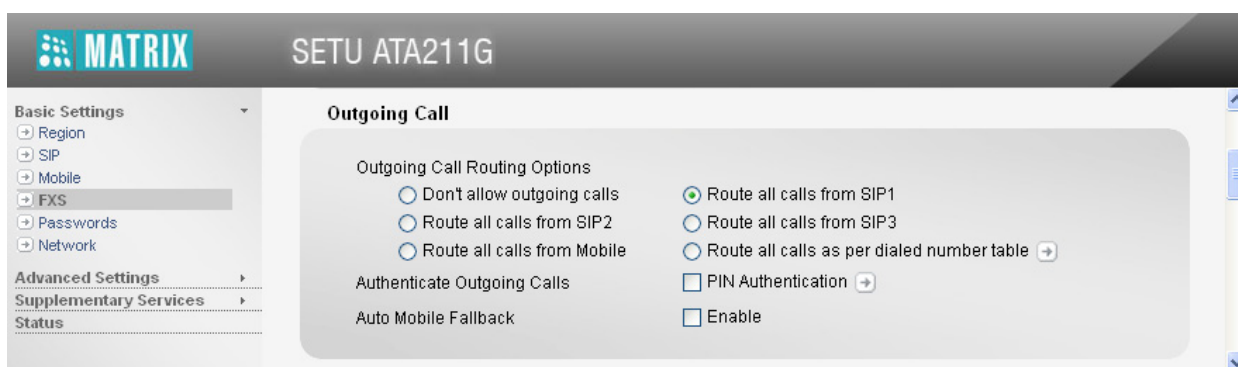
2. You can assign a **Name** to the FXS port, which will be displayed to the called party (if the instrument of the called party supports display name functionality).

The name you assign may consist of a maximum of 24 characters. Default: Blank

3. You can assign a **Number** to the FXS port. In case of Peer to Peer calls this number will be displayed to the called party.

The length of the number string may have a maximum of 24 characters. Default: Blank.

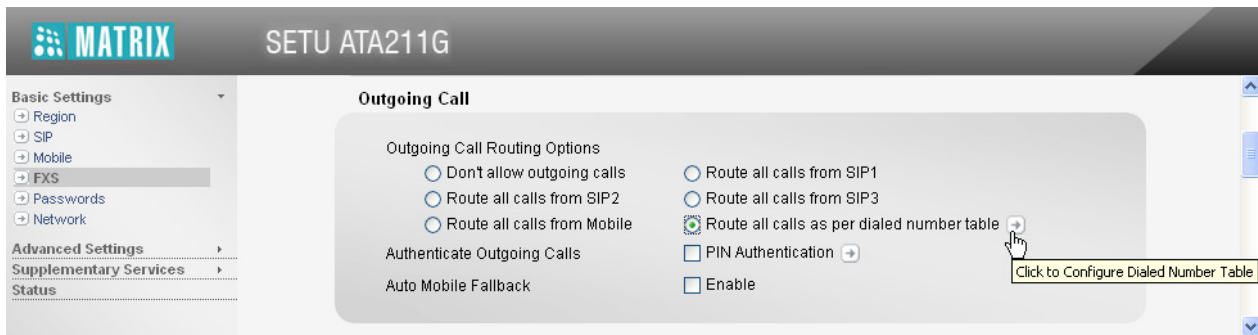
Outgoing Call



- Select an appropriate Outgoing Call Routing option on the FXS port. Default: SIP1.
- Select **Don't allow outgoing calls**, if you do not want to route calls through the FXS port.
- Select **Route all calls from SIP1**, if you want all outgoing calls from the FXS port to be routed through SIP Trunk 1.
- Select **Route all calls from SIP2**, if you want all outgoing calls from the FXS port to be routed through SIP Trunk 2.
- Select **Route all calls from SIP3**, if you want all outgoing calls from the FXS port to be routed through SIP Trunk 3.
- Select **Route all calls from Mobile**, if you want all outgoing calls from the FXS port to be routed through Mobile.
- Select **Route all calls as per Dialed Number Table**, if you want specific numbers to be routed through specific SIP Trunks only.

If you select this option, you must configure the Dialed Number Table. To do this,

- Click the arrow icon.



- The **Dialed Number Table** opens in a new window.

001-025 026-050 051-075 076-100

Dialed Number Table

Index	Number	Minimum Digits	Maximum Digits	Destination Route
001	0	01	24	SIP1
002	1	01	24	SIP1
003	2	01	24	SIP1
004	3	01	24	SIP1
005	4	01	24	SIP1
006	5	01	24	SIP1
007	6	01	24	SIP1
008	7	01	24	SIP1
009	8	01	24	SIP1
010	9	01	24	SIP1
011		01	24	SIP1
012		01	24	SIP1
013		01	24	SIP1
014		01	24	SIP1
015		01	24	SIP1
016		01	24	SIP1
017		01	24	SIP1

If dialed number does not match with either of the programmed number in dialed number table & call is made from FXS, route the call through

☐ None ☒ SIP1 ☐ SIP2 ☐ SIP3 ☐ Mobile

If dialed number does not match with either of the programmed number in dialed number table & call is made from Mobile, route the call through

☐ None ☒ SIP1 ☐ SIP2 ☐ SIP3 ☐ FXS

In this table, entries are stored at Index numbers from 001 to 100. For each index number,

- Enter the desired **Number** (max. 24 characters) you want to store in the table.
- For the number you entered, define the **Minimum Digits**, which the system should wait to receive before considering it as a valid number. Default: 01.
- For the number you entered, define the **Maximum Digits**, which the system should wait to receive before considering it as End of Dialing. Default: 24.
- For the number you entered, select as **Destination Trunk** the SIP Trunk the system should use to call the dialed number. Default: SIP1.

- At the bottom of the table, select Destination Trunks for dialed numbers that do not match with any of the entries configured in this table, i.e. the No Match Found.

For calls made from the FXS port, select the the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from FXS, route the call through.** Default, SIP Trunk 1.

For calls made from the FXS port, select the the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from Mobile, route the call through.** Default, SIP Trunk 1.

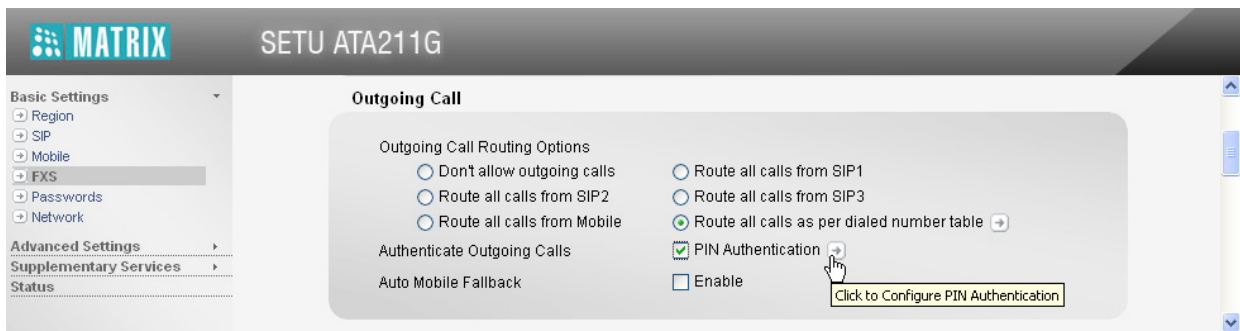
- Click **Submit** and close the window.

To know more, see [“Dialed Number Table”](#).

4. In **Authenticate Outgoing Calls** field, enable the **PIN Authentication** check box if you want ATA211G to allow outgoing calls from FXS port only after authenticating the user.

If you enable **PIN Authentication** check box, you must configure **PIN Authentication table**. To do this,

- Click the arrow.



- The **PIN Authentication** table opens in a new page.

PIN Authentication

Index	PIH	Index	PIH
001		051	
002		052	
003		053	
004		054	
005		055	
006		056	
007		057	
008		058	
009		059	
010		060	
011		061	
012		062	
013		063	
014		064	
015		065	
016		066	
017		067	
018		068	
019		069	
020		070	

Submit Default

- Enter the PIN to be authenticated against each Index in the Table.

The PIN number can be of a maximum of 4 digits. The characters allowed are (0-9, *, #).

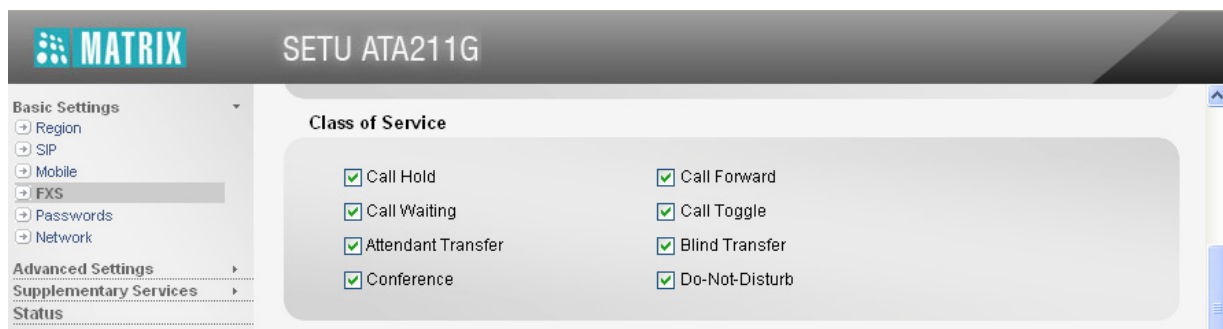
- If you have finished entering, click **Submit** to save.
- Close the window.

To know more about this feature, see [“PIN Authentication”](#).

5. Enable **Auto Mobile Fallback** check box if you want ATA to automatically route the calls through the Mobile port when the internet (ethernet) link is down and the SIP trunks cannot be used for call routing.

When you enable Auto Mobile Fallback, the system plays a beep for one second and routes the call to the Mobile port. If the Mobile port is disabled, busy, the call goes to error state.

Class of Service



- Select the features of SETU ATA211G in the **Class of Service²**(CoS) of the FXS Port .

SETU ATA211G offers the following set of features in Class of Service:

- [“Call Hold”](#)
- [“Call Waiting”](#)
- [“Call Transfer”](#)
- [“Conference”](#)
- [“Call Toggle”](#)
- [“Do-Not-Disturb”](#)
- [“Call Forward”](#)

By default all the features are enabled.

You may disable the feature you do not want to allow on the FXS Port.

2. *Class of Service (CoS) defines the set features of SETU ATA211G that the phone connected to the FXS port is to be allowed access to.*

Advanced

MATRIX SETU ATA211G

Basic Settings

- Region
- SIP
- Mobile
- FXS**
- Passwords
- Network

Advanced Settings

Supplementary Services

Status

Advance

Answer Signaling: ☐ None ☒ Polarity Reversal

Disconnect Signaling: ☐ None ☒ Polarity Reversal ☐ Open Loop Disconnect

Open Loop Disconnect Timer: 500 msec

Subscriber Type: ☒ Gateway ☐ Network

CLI Type: ☐ None ☐ DTMF ☐ FSK-BellCore ☒ FSK-ITU V.23

Flash Timer: 600 msec

Echo Cancellation: ☒ Enable

Tail Length: 32 msec

AC Impedance: 600 Ω

Speaking Volume: 0 db

Listening Volume: 0 db

1. Select the appropriate **Answer Signaling** Type on the FXS port.

Answer Signaling is required when SETU ATA211G has a PCO/Billing machine connected on its FXS port. When a call is made through SETU ATA211G and the called party answers the call, call maturity signal must be generated on the FXS port of SETU ATA211G so that the PCO/Billing device connected to its FXS port can start billing. In the absence of maturity information on the FXS port, the call will not be considered as matured, resulting in inaccurate billing.

- Select **None** if no answer signaling is to be generated on the FXS port.
- Select **Polarity Reversal** if answer signaling is to be generated in the form of Polarity Reversal on the FXS port.

Default: Polarity Reversal.

2. Select the appropriate **Disconnect Signaling** Type on the FXS port.

If a PCO machine is connected to the FXS port of the SETU ATA211G, whenever the Called Party (remote party) disconnects, the FXS port of SETU ATA211G needs to be informed of this event in the form of a signal, so that the PCO machine can consider the call as complete and stop billing. In the absence of such a signal, the call will be considered as completed only when the Caller goes On-hook, resulting in inaccurate billing.

- Select **None** if no signaling is to be generated on FXS for call disconnection.
- Select **Polarity Reversal** if call disconnection is to be signaled in the form of Polarity Reversal.
- Select **Open Loop Disconnect** if call disconnection is to be signaled in the form of Open Loop Disconnect signal.

If you select this option, you must configure the Open Loop Disconnect Timer.

Default: Polarity Reversal.

- **Open Loop Disconnect Timer:** If you selected **Open Loop Disconnect**, you may set the value of this timer, as required. Default: 500 milliseconds.

3. Select the **Subscriber Type** for SETU ATA211G. Default: Gateway.

When SETU ATA211G is interfaced with a service provider server—ITSP, the Matrix ETERNITY IP-PBX, or any other PBX—that supports supplementary services that require dialing of Flash, like Call Hold, Call Transfer, Call Waiting, etc., you must select the Subscriber Type for SETU ATA according to the extent of feature access you want on the FXS Port.

- Select **Network**, if you want to use the supplementary services supported by the PBX. When you set SETU ATA211G in the Network mode, you can access the service provider features by dialing flash. You will not be able to access the local features of SETU ATA211G.
- Select **Gateway**, if you want to use the supplementary features of SETU ATA211G primarily. In the Gateway mode, you will *also* be able to access the supplementary services of the service provider which requires the dialing of Flash. See [“Supplementary Services of Service Provider”](#).

4. Select the appropriate **CLI Type** according to the CLI Type supported by the telephone instrument connected to the FXS port.

SETU ATA211G supports three signaling protocols for CLI on the FXS port:

- DTMF
- FSK-ITU(V.23)
- FSK-BellCore

Default: FSK-BellCore

5. Configure the **Flash Timer**, as per your requirement. The Flash timer signifies the time period for which the loop current breaks. SETU ATA211G uses this event to activate various features such as Call Hold, Call Transfer, etc. Default: 600 msec.
6. To improve quality of speech, enable **Echo Cancellation**³. Default: Enabled.

Also define the **Tail Length** depending on the length of the echo. Valid Range: 8 msec to 32 msec. Default: 32 msec.

7. Select the appropriate **AC Impedance** according to the AC Impedance supported by the device connected to the FXS port of SETU ATA211G. The device may be a phone or a PBX. You may select from the following impedance options:
 - 600 Ω
 - 900 Ω
 - 270 Ω + (750 Ω || 150 nF)
 - 220 Ω + (820 Ω || 115 nF)
 - 370 Ω + (620 Ω || 310 nF)
 - 320 Ω + (1050 Ω || 230 nF)
 - 350 Ω + (1000 Ω || 210 nF)
 - 200 Ω + (680 Ω || 100 nF)Default = 600 Ω .

3. *Echo Cancellation is a technique that allows isolation and filtering of unwanted signals caused by echoes during speech.*

8. If required, you may adjust the **Speaking Volume** (Transmit) on the FXS port to increase or decrease the volume of your voice being transmitted to the remote party. Select the appropriate Speaking Volume from the drop-down list. Default: 0 db.
9. You may also adjust the **Listening Volume** (Receive) on FXS port to increase or decrease the volume of the remote party's voice being transmitted to you. Select the required Listening Volume level from the drop-down list. Default: 0 db.
10. Click **Submit** to save the settings.

Passwords

The system can be programmed by the Administrator or the User, by logging into Jeeves as **Admin** or **User** with their respective Passwords.

The default Admin and the default User Password are the same, **1234**.

Since the Admin Login gives you access to configure all the parameters of SETU ATA211G, while the User login has access to only user specific parameters, you may change the default password and use different Passwords for Admin and User Login.

The password must not exceed 4 digits. Only the digits 0 to 9 are allowed.

Changing Admin Password

You can change the Admin Password only from Jeeves.

1. Log in as **Admin** with your Password.
2. Click the **Basic Settings** link on the left pane to expand.
3. Click **Passwords**.



The screenshot shows the Jeeves web interface for SETU ATA211G. On the left is a navigation menu with 'Basic Settings' expanded, showing 'Region', 'SIP', 'Mobile', 'FXS', 'Passwords' (selected), and 'Network'. Below this are 'Advanced Settings', 'Supplementary Services', and 'Status'. The main content area is titled 'Admin Password' and contains three input fields: 'Old Password', 'New Password', and 'Re-Enter to Confirm'. Below this is a section for 'User Password'.

4. In the **Admin Password**,
 - Enter the current password in the **Old Password** field.
 - Enter the new password in the **New Password** field.
 - Type the new password again for confirmation in the **Re-Enter to Confirm** field.
5. Click **Submit** to save.



When you change the Admin Password, you will be logged out of Jeeves.

Restoring Default Admin Password

In case you forget the Admin password, you must set the password to the default value. To do this,

1. Lift the handset of your phone.

2. Dial #***

You will get dial tone after successful execution.

3. Replace handset.



*The Admin password will be set to the default value, if the **Default Admin Password when User defaults the ATA?** option is enabled in the **System Parameters**. To avoid this, make sure you disable this option in the [“System Parameters”](#).*

Changing User Password

You can change the User Password only from Jeeves.

To change User Password from Admin Mode:

1. Log in as **Admin**.
2. Click the **Basic Settings** link on the left pane to expand.
3. Click **Passwords**.

The screenshot shows the MATRIX SETU ATA211G web interface. On the left is a navigation pane with sections: Basic Settings (expanded), Advanced Settings, Supplementary Services, and Status. Under Basic Settings, the following links are visible: Region, SIP, Mobile, FXS, Passwords (highlighted), and Network. The main content area is titled 'Admin Password' and contains three input fields: 'Old Password', 'New Password', and 'Re-Enter to Confirm'. Below this is a section titled 'User Password' with two input fields: 'New Password' and 'Re-Enter to Confirm'. At the bottom right of the main area is a 'Submit' button with a checkmark icon.

4. In **User Password**,
 - Enter the new password in the **New Password** field.
 - Type the new password again for confirmation in the **Re-Enter to Confirm** field.
5. Click **Submit** to save.

To change User Password from User Mode:

1. Log in as **User**.
2. Click the **User Settings** link.

3. Click the **User Password**.



The screenshot shows the web interface of the MATRIX SETU ATA211G. On the left is a sidebar menu with 'User Settings' expanded, showing options like Supplementary Features, Speed Dialing, Network, User Password (selected), Syslog, Factory Defaults, and PCAP. Below this is a 'Status' section. The main content area is titled 'User Password' and contains three input fields: 'Old Password', 'New Password', and 'Re-Enter to Confirm'. A 'Submit' button with a checkmark icon is located at the bottom right of the form.

4. Enter the current password in the **Old Password** field.
5. Enter the new password in the **New Password** field.
6. Type the new password again for confirmation in the **Re-Enter to Confirm** field.

Restoring Default User Password

The User Password is automatically set to the default value whenever you reinstate factory defaults in the system. If you want to avoid having to set the entire system configuration to default settings, you may dial the related system command from your phone.

To restore default User Password,

1. Lift the handset of your phone.
2. Dial **#*****

You will get dial tone after successful execution.

3. Replace handset.



*Dialing this command string will also reset the Admin Password to factory default, if the **Default Admin Password when User defaults the ATA?** option is enabled in the System Parameters. To avoid this, make sure you disable this option before you default the User Password.*

Network Parameters

1. Click **Network**.

The screenshot shows the MATRIX SETU ATA211G web interface. The left sidebar contains a menu with 'Basic Settings' expanded, showing 'Region', 'SIP', 'Mobile', 'FXS', 'Passwords', and 'Network' (selected). Below this are 'Advanced Settings', 'Supplementary Services', and 'Status'. The main content area is titled 'Routing Mode' and contains three sections: 'Routing Mode' with radio buttons for 'NAT' (selected) and 'Bridge'; 'LAN' with input fields for 'IP Address' (192.168.2.21) and 'Subnet mask' (255.255.255.0); and 'WAN' with radio buttons for 'DHCP' and 'Static' (selected). Under 'Static', there are input fields for 'IP Address' (192.168.1.196), 'Subnet Mask' (255.255.255.0), 'Gateway', 'User ID', 'Password', and 'Service Name'. At the bottom, there is a 'DNS Server' field.

2. Click **More** button to view all Network parameters on this page.

Routing Mode

This screenshot is a zoomed-in view of the 'Routing Mode' section of the MATRIX SETU ATA211G web interface. It shows the 'Routing Mode' section with 'NAT' selected. Below it, the 'LAN' section shows the 'IP Address' as 192.168.2.21 and the 'Subnet mask' as 255.255.255.0. The 'WAN' section is partially visible but not fully shown.

- Select **Routing Mode**, according to your installation scenario as **NAT** or **Bridge**. Default: NAT.
- Select **NAT** as Routing Mode, if the LAN and WAN port of SETU ATA are in different network segments.

You will need to configure **NAT** as Routing Mode, when the SETU ATA211G is at the edge of the network and multiple hosts are connected behind ATA, and you want to share a common internet connection between all hosts.

- Select **Bridge** as Routing Mode, if the LAN and WAN port of SETU ATA are in the same network segment.

Selecting Bridge mode will disable Network Address Translation (NAT) on the WAN port.



When you set SETU ATA211G in Bridge Mode, you need not configure the LAN port IP Address and Subnet Mask. The WAN and the LAN port of SETU ATA211G can be accessed with the IP Address of the WAN port.

LAN

The LAN port of SETU ATA211G is used for connecting a standalone computer or a LAN Switch.

If you need to change the **IP Address** or the **Subnet Mask** of the LAN Port of SETU ATA211G, you may do it now.

Default IP Address of the LAN Port is: 192.168.2.21

Default Subnet Mask of the LAN Port is: 255.255.255.0

WAN

1. Select the **Connection Type** according to the IP addressing scheme of the network which SETU ATA211G is connected: DHCP, PPPoE, and Static.
 - Select **DHCP**, if the network uses a DHCP server to assign the IP address, Subnet Mask and Gateway address to SETU ATA211G.
 - Select **Static**, if you want to assign manually assign the IP address, Subnet Mask and Gateway address manually.

If you selected **Static**, enter the Static **IP Address** (default: 192.168.001.196) and the **Subnet Mask** (default: 255.255.255.0), and the **Gateway Address**.

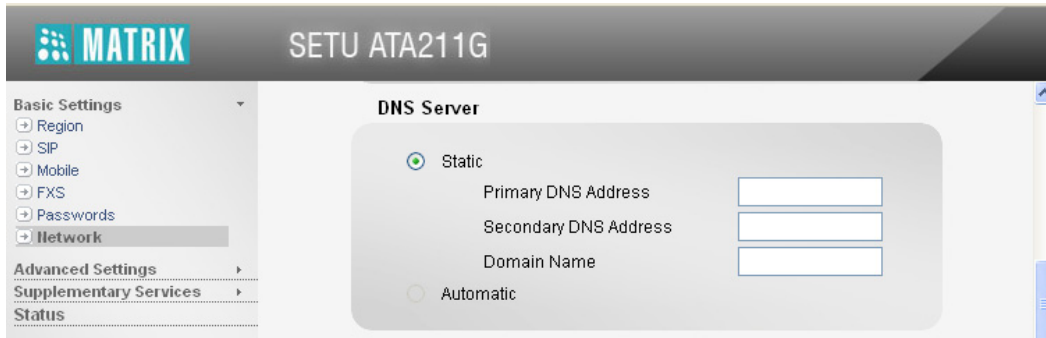
By default, the Connection Type for the WAN port is Static. IP Address: 192.168.001.196, Subnet Mask: 255.255.255.0.

- Select **PPPoE**, if the network uses PPPoE.

Enter **PPPoE User ID** (max. 40 characters) and **Password** (max. 24 characters). You may also enter the **PPPoE Service Name** (max. 24 characters), if provided.

DNS Server

DNS stands for Domain Name Server which is used to resolve domain name into IP address. You can select either **Static** or **Automatic**.



1. Select **Static** if you want to configure DNS manually.

- Enter **Primary DNS Address**.
- Enter **Secondary DNS Address**, if available.

Secondary DNS Address will be considered when the request to Primary DNS server fails.

- Enter the **DNS Domain Name**.

2. Select **Automatic** DNS if you want DNS Address and Domain name to be assigned by the DHCP/PPPoE server automatically.

This option will be applicable only if you have enabled DHCP or PPPoE as connection type.

Advanced

1. If the SETU ATA211G is connected in VLAN network, configure the **VLAN/CoS**.

This parameter enables the SETU ATA211G to add VLAN header to the packets generated by it. The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic⁴.

- Select the **VLAN/CoS** check box to enable VLAN ID tagging on all packets generated by the system. Default: Disabled.
- Enter the **VLAN ID** that you have assigned to the VLAN in which the SETU ATA211G is connected. The valid range for this is 0-4094. Default: 1.
- For **SIP CoS**, define the CoS (priority) bits which will be added in all SIP packets. The range of CoS bits is from 0 to 7. Default: 3.

4. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), i.e. better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- For **RTP CoS**, define the CoS (priority) bits which will be added in all RTP packets. The range of CoS bits is from 0 to 7. Default: 6.

2. SETU ATA211G will send all SIP messages using SIP QoS setting, enter the **SIP DiffServe/ ToS** as per your requirement. Valid range is from 00-63, Default: 26
3. SETU ATA211G will send all the RTP packets with RTP QoS setting, enter the **RTP DiffServe/ ToS** as per your requirement. Valid range is from 00-63, Default: 46
4. **MAC address** selection provides you two options: Unique and Clone. Default: Unique.

Select **Unique MAC Address**, if you want the system to use the unique MAC address assigned to it as source MAC address on all Ethernet frames.

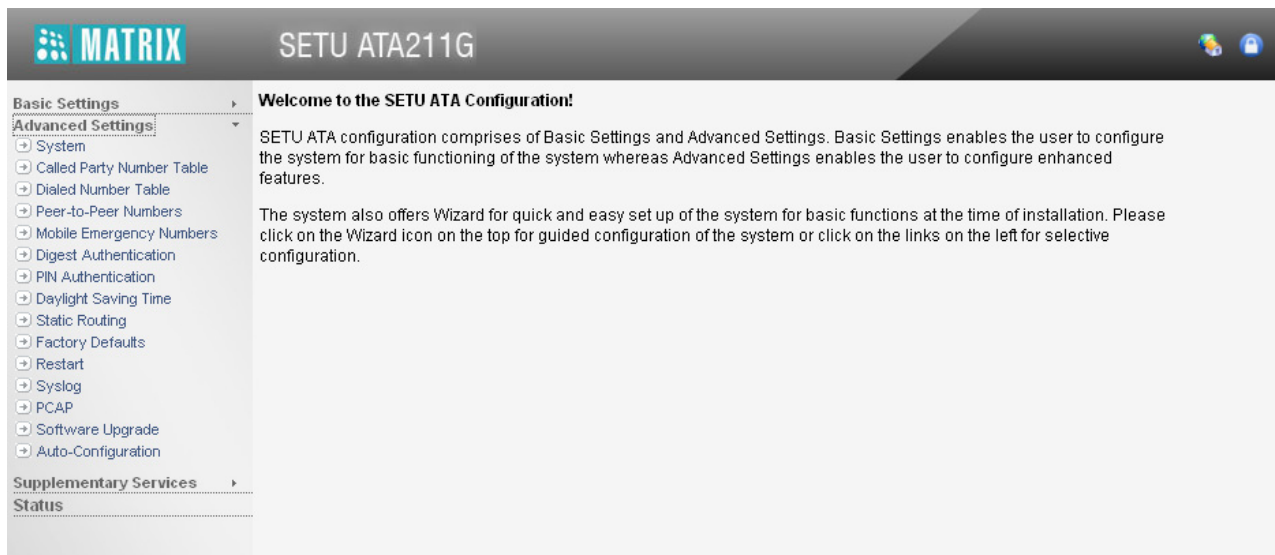
Select **Clone MAC Address**, if you want the system to use MAC address other than unique MAC address.

If you select Clone MAC Address, enter the desired MAC address manually in the **Clone MAC Address** field. MAC address is in hexadecimal format, e.g. 00:50:C2:55:B0:10.

5. SETU ATA211G has an embedded web server called Jeeves, for system configuration. You may change the **Web Server Port** as per your requirement. Valid range of the port is: 80, 1024-65535. Default: 80.
6. SETU ATA211G has an embedded FTP server for Software Upgrade. You may change the **FTP Server Port** as per your requirement. Valid range of the port is: 21, 1024-65535. Default: 21.
7. After you have set all the parameters as per your requirement, click **Submit** at the bottom of the page to save your Network settings.
8. You will get a prompt, 'This will restart the system, do you want to continue?'
9. Click **OK**. The system will restart.

The **Advanced Settings** include the features and facilities listed below::

- “System Parameters”
- “Called Party Number Table”
- “Dialed Number Table”
- “Peer-to-Peer Numbers”
- “Mobile Emergency Numbers”
- “Digest Authentication”
- “Daylight Saving Time”
- “Static Routing”
- “Reinstate Factory Defaults”
- “Restart SETU ATA211G”
- “Syslog”
- “PCAP”
- “Software Upgradation”
- “Auto Configuration”



System Parameters

Certain parameters of SETU ATA211G are applied system-wide and are not specific to a port. These parameters are called System Parameters.

To configure System Parameters,

1. Log into Jeeves as **Admin** with your password.
2. Click the **Advanced Settings** link on the left pane to expand the links.
3. Click **System**. The System Parameters page opens.
4. Click the **More** button to view all the System parameters on this page. You may configure the following parameters on this page, as required:

General

The screenshot shows the 'General' configuration page for the SETU ATA211G device. The left sidebar contains a navigation menu with 'Basic Settings', 'Advanced Settings' (expanded), 'System' (selected), 'Dial Party Number Table', 'Dialed Number Table', 'Peer-to-Peer Numbers', 'Mobile Emergency Numbers', 'Digest Authentication', 'PIN Authentication', 'Daylight Saving Time', 'Static Routing', 'Factory Defaults', 'Restart', 'Syslog', 'PCAP', 'Software Upgrade', 'Auto-Configuration', 'Supplementary Services', and 'Status'. The main content area is titled 'General' and contains the following parameters:

Parameter	Value	Unit
System Name	atrix SETU ATA211G	
First Digit Wait Timer	15	sec
Inter Digit Wait Timer	5	sec
Transfer Notification Timer	60	sec
Play Routing Tone	☒ Enable	
Call Release Timer	☒ Enable	
Timer	999	min
VoIP Silence Disconnect Timer	☒ Enable	
Timer	99	min
RCOC Record Delete Timer	999	min
	<button>Clear RCOC Table</button>	
Forward Error Correction	☐ Enable	
Voice Activity Detection	☐ Enable	
Default Admin Password when User Defaults the ATA	☒ Yes	

- **System Name:** You can assign a name to SETU ATA211G, as 'System Name'. This name has significance when multiple SETU ATA211G are connected in the same LAN network.

The Name you assign may contain a maximum of 24 characters. Default: Matrix SETU ATA211G.

- **First Digit Wait Timer:** The First Digit Wait Timer signifies the time for which the system waits for receiving the first digit after going Off-hook from the FXS Port. On expiry of this timer, the system will give error tone to the user. Valid range: 01 to 99 seconds. Default: 15 seconds.
- **Inter Digit Wait Timer:** Inter Digit Wait Timer signifies the time period for which the system should wait between two consecutive digits when receiving the digits from the FXS Port. On expiry of this timer, the number is considered to be complete and is dialed. Refer [“Dialed Number Table”](#) topic for information

about End of Dialing while dialing minimum or maximum numbers. Valid range: 01 to 99 seconds. Default: 5 seconds.

- **Transfer Notification Timer:** This is the time for which the system will wait for notification of the status of a transferred call, i.e. whether the transfer target is busy, has answered, has disconnected, etc. Refer "[Call Transfer](#)" topic to know more. Valid Range: 01-99 secs. Default: 60 seconds.
- **Play Routing Tone:** By default, when calls are routed from Mobile and SIP trunk and no response is received from the called party, the system Plays Routing Tone to the caller.

You may clear the **Play Routing Tone** check box, if you do not want the system to play routing tone.

- **Call Release Timer:** This Timer may also be used to restrict the duration of calls made from the FXS port, Mobile Port and SIP trunks. This timer allows the system to release the matured calls made from the FXS port, Mobile Port and from SIP trunks after a specified time period.

The Call Release Timer is used for disconnecting calls that remain connected in certain conditions. For example, the FXS port has disconnected the call, but no disconnect event is received from the SIP trunk. In this case, the call remains live in the system. On the expiry of the Call Release Timer, this call will be disconnected.

To apply this timer, select the **Call Release Timer** check box and set the **Timer** to the required duration from 001 to 999 minutes. Default: 999 Minutes.

- **VoIP Silence Disconnect Timer:** This feature allows the system to disconnect the call if silence is detected during speech, for more than the specified duration.

To apply this timer, select the **VoIP Silence Disconnect Timer** check box and set the **Timer** to the required duration from 001 to 99 minutes. Default: 99 Minutes

- **RCOC Record Delete Timer:** You can delete the RCOC records either automatically by specifying time duration in the field 'RCOC Record Delete Timer'. Valid range: 1 to 999 minutes. Default: 999 minutes.

You can also delete the RCOC records any time you want by clicking the '**Clear RCOC Table**' button.

- **Forward Error Correction:** Forward Error Correction is used for data transmission whereby ATA211G adds redundant data to its messages, which allows the receiver to detect and correct errors without asking the sender for additional data.

Select the **Forward Error Correction** check box to enable. Default: Disabled.

- **Voice Activity Detection:** Voice Activity Detection is a software application which allows data network, carrying voice traffic over the internet, to detect the absence of audio and conserve bandwidth by preventing the transmission of **Silent Packets** over the network.

Select the **Voice Activity Detection** check box if you want SETU ATA211G to apply Voice Activity Detection (also called Silence Suppression). Default: Disabled.

- **Default 'Admin' password when 'User' defaults the ATA?:** By default, the Admin Password will be set to its default value, when the User sets the system to default.

To keep the Admin Password unchanged, disable this flag. Default: Enabled.



If you are an ITSP, you may use this flag (disable) to prevent your customers (end users) from setting the Admin Password to default and gaining unrestricted access to system configuration.

NAT

The screenshot shows the Matrix SETU ATA211G configuration interface. On the left is a sidebar with a tree view containing 'Basic Settings', 'Advanced Settings' (with sub-items like System, Called Party Number Table, etc.), 'Supplementary Services', and 'Status'. The main area is titled 'NAT' and contains the following settings:

- STUN Server Address:Port**: A text input field followed by a port field set to '03478'.
- SIP Port Fetched Using STUN**: A checkbox labeled 'Enable' which is checked.
- Router Public IP Address**: A text input field.
- UDP NAT Keep Alive**: A checkbox labeled 'Enable' which is unchecked.
- Keep Alive Message**: Radio buttons for 'NOTIFY' (selected) and 'REGISTER'.
- Interval**: A spinner box set to '120' with 'sec' as the unit.
- TCP NAT Keep Alive**: A checkbox labeled 'Enable' which is unchecked.
- Interval**: A spinner box set to '120' with 'sec' as the unit.

Below the NAT section is a 'SIP' section with the following setting:

- 100rel/PRACK**: A checkbox labeled 'Enable' which is unchecked.

- STUN (Simple Traversal of UDP through NAT) server facilitates traversing through most NATs, except symmetric NATs. So, if your router has Symmetric NAT, do not configure STUN.

If your SETU ATA211G is located behind a NAT router that is other than symmetric,

- In the **STUN Server Address: Port** field, enter the STUN Server Address and the Listening Port of the STUN Server.

The STUN Server Address can have a maximum of 40 characters.

The valid range of the STUN Server Port is from 1024–65535. Default: 3478.

- Clear the **SIP Port fetched using STUN** check box, if ATA211G is located behind the NAT router and you have forwarded the SIP listening port of the ATA211G in the router.

Keep the **SIP Port fetched using STUN** check box enabled, if you have *not* forwarded the SIP Listening Port in the router.



Make sure you configure the **NAT Type** on the SIP Trunk as **STUN**. See [“SIP Trunks”](#).

- **Router's Public IP Address:** Router's public IP address specifies the public IP address of the NAT router behind which system is located. Default: Blank.

You need to configure this field only if the system is located behind the NAT router and a Static IP Address is assigned as Public IP Address of the Router.



Make sure you configure the **NAT Type** on the SIP Trunk as **Router's IP Address**. See [“SIP Trunks”](#).

- **UDP NAT Keep Alive:** When SETU ATA211G is connected behind a NAT router and SIP messages are transported over UDP, NAT Keep Alive messages must be sent to refresh the binding in the NAT router.

Select the **UDP NAT Keep Alive** check box to enable. Default: Disabled.

Select the the type of **Keep Alive Message** to be sent. You may select either REGISTER or NOTIFY. Default: NOTIFY.

As **Interval**, set the time period after which the system should send Keep Alive messages. This time period should be less than the NAT binding timer of the router. The valid range for the UDP NAT Keep Alive Interval is 001–999 seconds. Default: 120 seconds.

- **TCP NAT Keep Alive:** When SETU ATA211G is connected behind a NAT router, and SIP messages are transported over TCP, NAT Keep Alive messages must be sent to refresh the binding in the NAT router.

Select the **TCP NAT Keep Alive** check box, if you want the system to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: disabled.

As **Interval**, set the time period after which the system should send Keep Alive messages. This time period should be less than the NAT binding timer of the router. The valid range for the TCP NAT Keep Alive Interval is 001–999 seconds. Default: 120 seconds.

SIP

MATRIX SETU ATA211G

Basic Settings

Advanced Settings

- System
- Called Party Number Table
- Dialed Number Table
- Peer-to-Peer Numbers
- Mobile Emergency Numbers
- Digest Authentication
- PIN Authentication
- Daylight Saving Time
- Static Routing
- Factory Defaults
- Restart
- Syslog
- PCAP
- Software Upgrade
- Auto-Configuration

Supplementary Services

Status

SIP

100rel/PRACK	☐ Enable
SIP over TCP	☒ Enable
SIP UDP Port	05060
SIP TCP Port	05060
RTP Listening Port	08000
SIP INVITE Timer	030 sec
SIP Provisional Timer	060 sec
General Request Timer	20 sec

- **100rel/PRACK:** This parameter is to be configured if you want to support reliable transmission of (SIP) provisional responses.

Select the **100rel/PRACK Enable** check box, if you want the SETU ATA211G to use 100rel SIP extension for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgement). Default: Disabled.

- **SIP Over TCP:** SETU ATA211G supports transporting of SIP messages over User Datagram Protocol (UDP) as well as Transfer Control Protocol (TCP) connection. Despite the advantages that SIP over TCP offers, it is more common to use UDP to transport SIP messages.

By default, SIP over TCP is enabled. If you want to receive SIP messages over TCP keep this option enabled.

You must also enable **TCP** or **TCP (Fallback to UDP)** on the SIP Trunk.

- **SIP UDP Port:** This is the port on which the SETU ATA211G listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1024–65534. Default: 5060.
- **SIP TCP Port:** This is the port on which the SETU ATA211G listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025–65534. Default: 5060.
- **RTP Listening Port:** This is the port on which the SETU ATA211G listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1024–65526. Default: 8000.
- **SIP INVITE Timer:** This is the time in seconds for which SETU ATA211G waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the SETU ATA211G terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 010–180 seconds. Default: 30 seconds.
- **SIP Provisional Timer:** This is the time in seconds for which SETU ATA211G waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the SETU ATA211G terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 010–180 seconds. Default: 20 seconds.
- **General Request Timer:** This is the time in seconds for which the SETU ATA211G waits for response of a transaction request. This timer starts on initiating a transaction and stops on the receipt of a response for the request. On expiry of the timer, the SETU ATA211G clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10–60 seconds. Default: 20 seconds.
- Click **Submit** to save the settings.

Called Party Number Table

Called Party Number Table is used for routing incoming calls on the SIP trunks on the basis of the Called Party Number received, without answering the call.

You must configure this table, if you have selected '**Route as per called party number**' as the *Incoming Call Routing* option for the SIP trunk.

In the Called Party Number Table, you can store a maximum of 10 numbers.

Each number is stored at an Index in the table. For each number that you enter in the table, you must define the minimum and maximum digit length and select the desired port—FXS or Mobile—to route the call.

Consider the following example of the Called Party Number Table:

Index	Number	Minimum Digit	Maximum Digit	Destination Route
01	(No match found)	01	24	Mobile
02	2	7	7	Mobile
03	3	7	7	Mobile
04	98	10	10	FXS
05	00	12	12	FXS

In this Table:

- The Called Party Number strings received in the incoming call are stored in the Number column against Index 02 to 10. The first Index is reserved for the No Match Found Entry. The numbers may be prefixes or the complete number strings.
- For each number, the digit length is defined as Minimum Digit and Maximum Digit.
- For each number, the port to be used for routing the call is selected as the Destination Route.
- The incoming calls received from local numbers are to be routed through the Mobile port. These are 7 digit numbers starting with '2' or '3'. So, the prefixes of these numbers, '2' and '3' are entered in the Number field. The Minimum and Maximum Digits are defined as '7'. The Destination Route is selected as Mobile port.
- Incoming calls from mobile numbers starting with '98' are to be routed to the FXS port. Since the mobile numbers are 10 digit numbers, the Minimum and Maximum digits are configured as '10' and the Destination Route is selected as FXS.
- Twelve-digit numbers starting with '00', are to be routed through the FXS port. So, the Minimum and Maximum digits are configured as '12' and the Destination Route is selected as FXS.

When there is an incoming call and the incoming call routing option is selected as **Route as per called party number**, the system will check the Called Party Number.

It will use the best match logic to match the number with the table. If the number received starts with '2', the system will match with the entry at Index02. The system will match the length of the digits with the minimum and maximum configured for the entry at Index02. If the minimum and maximum digits match the system routes the call through the Mobile port selected as Destination Route for numbers starting with '2'.

To be considered as a valid number, the number received must be equal to or more than the Minimum digits configured. If the number received matches the Minimum digits but exceeds the maximum digits, the system will strip off the extra digits and route the number.

If the incoming CLI does not match with any entry in Called Party Number Table (i.e. the number does not exist in the table, the call will be routed through the destination route for the first entry at Index 001, i.e. the No Match Found entry, in the table.

To configure Called Party Number Table,

1. Log in as **Admin** with your password.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Called Party Number Table**.
The Called Party Number Table page opens.

Index	Number	Minimum Digits	Maximum Digits	Destination Route
01	No Match Found	01	24	None
02		01	24	Mobile
03		01	24	Mobile
04		01	24	Mobile
05		01	24	Mobile
06		01	24	Mobile
07		01	24	Mobile
08		01	24	Mobile
09		01	24	Mobile
10		01	24	Mobile

4. For each Index number,
 - Enter the desired **Number** you want to store in the table. The number must not exceed 24 characters.
The first Index number 01, is reserved for No Match Found. Start your entries from Index 02.
 - For the number you entered, define the **Minimum Digits**, which the system should wait to receive before considering it as a valid number. Default: 01.
 - For the number you entered, define the **Maximum Digits**, which the system should wait to receive before considering it as a valid number. Default: 24.
 - For the number you entered, select required **Destination Route**, the system should use to route the incoming call. Default: None for Index 01 and Mobile for Index 02 to 10.
5. Click **Submit** save your entries.

Dialed Number Table

Dialed Number Table allows the SETU ATA211G to select the most suitable trunk for outgoing calls on the basis of the number dialed.

You must configure this table, if you have selected '**Dialed number based routing**' for the FXS Port and the Mobile Port.

The Dialed Number Table is a list of numbers or part numbers (which you must configure), with a preferred destination port/trunk for each number: FXS, SIP1, SIP2, SIP3, or Mobile. When the user dials a number, SETU ATA211G checks the Dialed Number Table for a match using the best fit logic. If a match is found, it uses the port/trunk selected for the number in the table to route the call.

In the Dialed Number table, up to 100 numbers can be stored.

Each number is stored at an Index in the table. For each number that you enter in the table, you must define:

- the minimum number of digits that the system should wait to receive to consider it a valid number to be dialed out.
- the maximum number of digits which the system should wait to receive to consider it as “End of Dialing” and dial out the number.
- the Destination Route (SIP Trunk, Mobile port or FXS port), which the system should use to dial out the number.



The Dialed Number Table is to be configured only if you have selected:

- the **Outgoing Call Routing** option for the FXS port as **Route all calls as per Dialed Number Table**.
- the **Incoming Call Route** for the Mobile port as **Dialed Number Table**.

When a number is dialed from the FXS port or received on the Mobile port,

- SETU ATA211G will match the dialed number with the entries of the Dialed Number Table.
- If a match is found, it will check the Minimum Digits programmed for this entry to consider it a valid number.

If the length of the number string you dialed matches with the Minimum Digits programmed for this entry, and “End of Dialing” is detected⁵, the system will consider it as a valid number. It will not wait to match it with the Maximum Digits you configured for this entry. The system will dial out the number using the Destination Route you have selected for this entry in the table.

However, if the number of digits dialed by you are less than the Minimum Digits you configured, you will get error tone.

- If the dialed number string exceeds the Minimum Digits, but is less than the Maximum Digits, the system will check the Maximum Digits programmed for this entry.
- If the length of the dialed number string equals the Maximum Digits programmed for this entry, the system will not wait for further digits to be dialed. It will consider the Maximum Digits defined as End of Dialing, and will dial out the number using the Destination Route you selected for this entry in the table.

5. You pressed the # key or the Inter Digit Wait Timer has expired. See “End of Dialing”.

- You can also configure the Destination Route for dialed numbers that do not match with the table by the Source Port.
 - If the Mobile port is the Source Port, you can route the call through — SIP1/SIP2/SIP3 or FXS.
 - If FXS port is the Source Port, you can route the call through— SIP1/SIP2/SIP3, or Mobile.
- You can also restrict dialed numbers that do not match with the table by not selecting any destination route.

Configuring Dialed Number Table

If you have not already configured the Dialed Number Table for the FXS and Mobile port, you may do so now.

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Dialed Number Table**. The Dialed Number Table page opens.

MATRIX SETU ATA211G

001-025 026-050 051-075 076-100

Dialed Number Table

Index	Number	Minimum Digits	Maximum Digits	Destination Route
001	0	01	24	SIP1
002	1	01	24	SIP1
003	2	01	24	SIP1
004	3	01	24	SIP1
005	4	01	24	SIP1
006	5	01	24	SIP1
007	6	01	24	SIP1
008	7	01	24	SIP1
009	8	01	24	SIP1
010	9	01	24	SIP1
011		01	24	SIP1
012		01	24	SIP1
013		01	24	SIP1
014		01	24	SIP1
015		01	24	SIP1
016		01	24	SIP1
017		01	24	SIP1

Index	Number	Minimum Digits	Maximum Digits	Destination Route
017		01	24	SIP1
018		01	24	SIP1
019		01	24	SIP1
020		01	24	SIP1
021		01	24	SIP1
022		01	24	SIP1
023		01	24	SIP1
024		01	24	SIP1
025		01	24	SIP1

If dialed number does not match with either of the programmed number in dialed number table & call is made from FXS, route the call through

☐ None ☒ SIP1 ☐ SIP2 ☐ SIP3 ☐ Mobile

If dialed number does not match with either of the programmed number in dialed number table & call is made from Mobile, route the call through

☐ None ☒ SIP1 ☐ SIP2 ☐ SIP3 ☐ FXS

4. For each Index number,

- Enter the desired **Number** you want to store in the table. The number must not exceed 24 characters.
- For the number you entered, define the **Minimum Digits**, which the system should wait to receive before considering it as a valid number. Default: 01.
- For the number you entered, define the **Maximum Digits**, which the system should wait to receive before considering it as End of Dialing. Default: 24.
- For the number you entered, select as **Destination Route** the port the system should use to call the dialed number. Default: SIP 1
- At the bottom of the table, select the Destination Trunk for dialed numbers that do not match with any of the entries configured in this table (No Match Found) by the Source Port.
 - For calls made from the FXS port, select the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from FXS, route the call through**. Default, SIP Trunk 1.

If you do not want to route the dialed numbers from the FXS port that do not match with the table, select **None**. The System will reject the call.

- For calls made from the Mobile port, select the destination trunk for the option **If dialed number does not match with either of the programmed number in the Dialed Number Table & call is made from Mobile, route the call through**. Default, SIP Trunk 1.

If you do not want to route the dialed numbers from the Mobile port that do not match with the table, select **None**. The System will reject the call.

5. Click **Submit** to save your entries.

- For the number you entered, in the **Destination Address** field in the table, enter the **IP Address** of the WAN Port of ATA connected at Location B. In this case, 190.1.1.100
- For the number you entered, configure the digit length in the table. As the number you want to dial, 3001, has four digits, configure **Minimum Digit** for this number as '4'. Also configure the **Maximum Digit** as '4'.

The Peer-to-Peer table you configure for SETU ATA211G at Location-A would look like this:

Index	Number	Name	Minimum Digit	Maximum Digit	Destination Address
001	No Match Found		01	24	
002	3001		04	04	190.1.1.100
003					
004					



As the system uses the best match logic to match number strings in the Peer-to-Peer table, you may configure only the prefix of the number to be dialed. Instead of the complete number string, you may configure only the prefix as follows, the system will place all calls that start with '3' to the IP Address 190.1.1.100

Index	Number	Name	Minimum Digit	Maximum Digit	Destination Address
001	No Match Found		01	24	
002	3		04	04	190.1.1.100
003					
004					

- Similarly, the Peer-to-Peer table you configure at Location-B should look like this:

Index	Number	Name	Minimum Digit	Maximum Digit	Destination Address
001	No Match Found		01	24	
002	2001		04	04	190.1.1.200
003					
004					

- When user 2001 of Location-A or a caller on the Mobile port calls 3001, the call is placed to the IP address 190.1.1.100 as the system finds a best match entry for the dialed number at Index 002 of the Peer-to-Peer table.
- On receiving a call, the SETU ATA at Location-B routes this call to the FXS port, 3001.
- Similarly, when the user 3001 of Location B or a caller on the Mobile port, calls 2001, the call is placed to the IP address 190.1.1.200 as the system finds a best match entry for the dialed number at Index 002 of the Peer-to-Peer table.

Configuring Peer-to-Peer Calling

To use Peer-to-Peer calling, do the following:

- enable the SIP trunk.
- set the SIP Trunk Mode to **Peer-to-Peer**.
- configure the number callers must dial to reach this SIP trunk as its SIP ID.
- set the Incoming Call Routing option on the SIP trunk as **Route to FXS**.
- configure Outgoing Call Route on the FXS port.
- configure Incoming Call Routing on the Mobile port.
- configure the Peer-to-Peer table.

If you have not configured the Peer-to-Peer table from the **SIP Trunks** page under *Basic Settings*, you may do so now.

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Peer-to-Peer Numbers**. The Peer-to-Peer Table opens.

MATRIX SETU ATA211G

Basic Settings
Advanced Settings
System
Called Party Number Table
Dialed Number Table
Peer-to-Peer Numbers
Mobile Emergency Numbers
Digest Authentication
PIN Authentication
Daylight Saving Time
Static Routing
Factory Defaults
Restart
Syslog
PCAP
Software Upgrade
Auto-Configuration
Supplementary Services
Status

001-050 051-100 101-150 151-200 201-250 251-300 301-350 351-400 401-450 451-500

Peer-to-Peer Numbers

Index	Number	Name	Minimum Digits	Maximum Digits	Destination Address
001	No Match Found		01	24	
002			01	24	
003			01	24	
004			01	24	
005			01	24	
006			01	24	
007			01	24	
008			01	24	
009			01	24	
010			01	24	
011			01	24	
012			01	24	
013			01	24	
014			01	24	
015			01	24	
016			01	24	

Submit Default

In the Peer-to-Peer table, each entry is stored against an Index Number from 001 to 500. The first entry at Index001 is reserved for No Match Found.

4. For each Index number,
 - In the **Number** field, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 24 characters. Default: Blank.

If the number to be dialed out is <dialednumber@destination address>, for example, 1234@abc.com, you must enter 1234 in this field.

- In the **Name** field, enter a name to identify the number string you configured. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.

The name you configure here will not be used in SIP signaling.

- As **Minimum Digits**, define the minimum length of the number string that must be dialed for the system to consider it as a valid number. Default: 01.
- As **Maximum Digits**, define the maximum length of the number string that must be dialed out for the system to consider it the complete number string. Default: 24.
- In the **Destination Address** field, enter the domain name or IP Address to where the call is to be placed. The Destination Address may consist of up to 40 characters (maximum). Default: Blank.

For example, if the peer-to-peer number to be dialed out is 1234@abc.com, enter abc.com as Destination Address. If the number is 1234@192.168.1.197, enter 192.168.1.197 as the Destination Address. The Destination Address can also be in the form of Address: Port number.

5. Click **Submit** to save your entries.

You may log out of Jeeves.

Making Peer-to-Peer Calls

To make a Peer-to-Peer Call:

- Go Off-hook.
- Dial the IP Address of the remote party or the number of the remote party stored in the Peer-to-Peer Table.
- To dial an IP Address, press * key (star/asterisk).

For example, to call the IP Address 192.168.200.200, dial 192*168*200*200.

When you press the * key, it is interpreted by the SETU ATA211G as a '.' (dot/period).

- Talk, when the remote party answers the call.

Mobile Emergency Numbers

SETU ATA211G supports **Mobile Emergency Numbers** feature which allows you to make calls to emergency services such as Ambulance, Fire Brigade, Police, etc from its Mobile port. Dialing Emergency Numbers from the mobile port is possible even if the SIM card is not inserted in the Mobile port or even when the module is not registered with the GSM network.

The Mobile Emergency Number feature works on the basis of the Emergency Number Table. You must configure this table in Jeeves. The Emergency Number Table supports five entries where only first two entries are configurable and the last three entries are fixed.

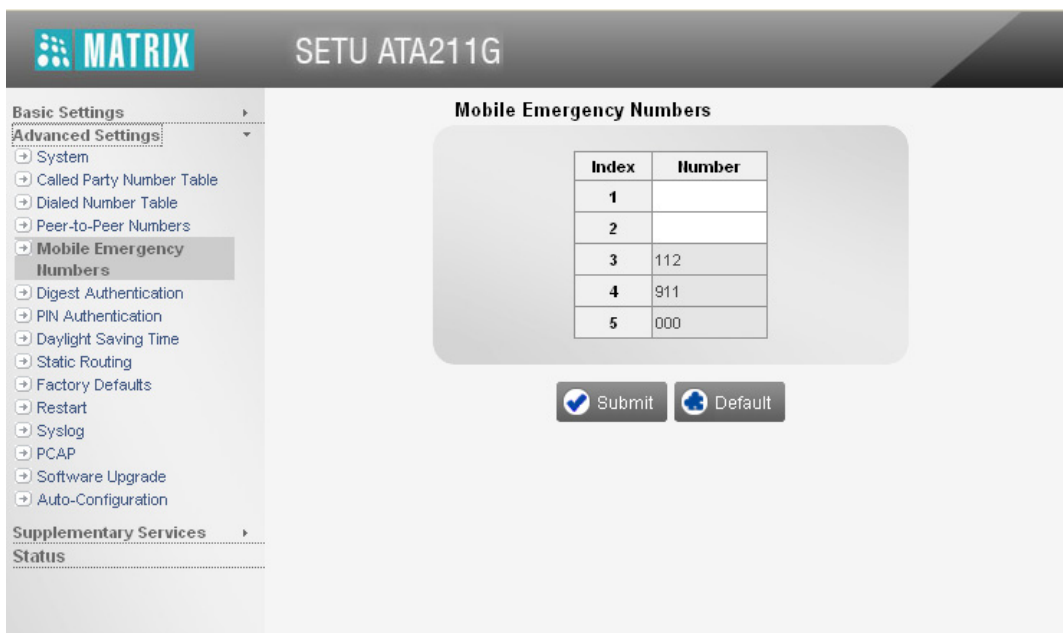
Each Emergency Number is stored at an Index in the table. Each Emergency Number is of maximum 3 digits.

- If you dial a number, the system compares the number with the numbers configured in the **Number** column of the Emergency Number Table.
- The system uses the **perfect match** logic for matching the dialed number with the configured Emergency Numbers.
- If the dialed number matches perfectly with the configured Emergency Numbers in the table, then ATA211G does not wait for the end of dialing and outdials the number automatically.
- If the dialed number does not match perfectly with the configured Emergency Number and the end of dialing is received, then the number is dialed as per outgoing logic.

Configuring Emergency Number Table

To configure Emergency Number Table,

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Mobile Emergency Numbers**. The Mobile Emergency Number Table opens.



4. For each Index entry, configure the Emergency Number in the **Number** column. You can configure each Emergency Number of maximum 3 digits and 0-9 digits are allowed.

The default Emergency Number Table is as below:

Index	Emergency Number
1	Blank
2	Blank
3	112
4	911
5	000



- *Avoid configuring conflicting numbers as Emergency Numbers.*
- *Emergency Numbers have priority over PIN Numbers and Access Codes.*
- *The system dials out the Emergency Numbers without waiting for End-of-Dialing.*

Digest Authentication

Digest Authentication is a challenge-based authentication service of SIP to authenticate the identity of the originator of SIP request in the INVITE message. The recipient of the request can ascertain whether or not the originator of the request is authorised to make the request. When the digest credentials of the originator - User Name and Password - in the INVITE message are authenticated and accepted by the recipient, the originator and the recipient are connected.

SETU ATA211G supports Digest Authentication. The Digest Authentication feature works on the basis of the Digest Authentication Table, in which the credentials, namely the User Name and Passwords of trusted/authorised calling party SIP devices are stored. You must configure this table.

When you enable this feature on a SIP trunk, for all incoming calls (SIP requests),

- SETU ATA211G will challenge the identity of the calling party, i.e. the SIP device initiating the request to send its digest credentials.
- When the calling party sends its credentials, SETU ATA211G authenticates the credentials by matching it with its Digest Authentication Table.
- If a match is found, the calling party will be authenticated and the call will be allowed on the SIP trunk.
- If no match is found, SETU ATA211G will consider it as invalid authentication information and reject the call.

You may use Digest Authentication to

- restrict access to SETU ATA211G to specific callers.
- prevent unwanted or malicious calls.

Let us understand Digest Authentication with the help of an example:

- A Company has its head office in Mumbai and branch offices in the cities of Kolkata, Chennai and New Delhi.
- For voice communication over IP, the Company has installed SETU ATA211G in all its branches for making Peer-to-Peer Calls. (For more details, refer [“Peer-to-Peer Numbers”](#) topic).
- The Company wants to use a SIP trunk for exclusively for inter-office calling.
- To be able to do this, the Company must do the following in all its branch offices:
 - set the SIP Trunk mode of the desired SIP trunk to Peer-to-Peer mode, and configure Peer-to-Peer Numbers⁶.
 - enable Digest Authentication on this SIP Trunk.
 - configure the Digest Authentication Table with the User Name and Password of the SIP devices of all branches, from which calls are to be allowed on this SIP trunk. In this case, the User Name and Password will be of the dedicated Peer-to-Peer SIP trunk in the branch offices. For example, in the Digest Authentication Table, you configure at the Mumbai office, you must configure the User Name

6. Static IP address is assigned to the WAN port of all Gateways and the SIP trunks are set to Peer-to-Peer mode in all the offices.

and Password for the dedicated SIP trunks at the offices in Kolkatta, Chennai and New Delhi. Similarly, in the Digest Authentication Table you configure in the New Delhi office, you must configure the User Name and Password for the dedicated SIP trunks at the offices in Mumbai, Kolkatta, Chennai.

- With Digest Authentication configured at all branches, whenever an incoming call is received on the SIP trunk, the SIP trunk will challenge the identity of the caller. When the digest credentials sent by the calling device matches with the Digest Authentication Table, the call will be allowed on the SIP trunk.
- If the digest credentials do not match, the call will be rejected.
- Thus, the Company can use Digest Authentication to restrict all call except those from its branch offices on a SIP trunk.

Configuring Digest Authentication

To use this feature, make sure you have enabled **Digest Authentication** on the desired SIP Trunk and configure the Digest Authentication Table.

If you have not already configured the Digest Authentication Table on the SIP Trunk parameters page, you may do so now.

1. Log in as **Admin** with your Admin Password.
2. Click the **Advanced Settings** link on the left pane to expand the links.
3. Click **Digest Authentication**.

Index	User ID	User Password
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		

Submit Default

The Digest Authentication Table page opens. You can configure upto 500 entries in this table. This Table is common for all SIP trunks.

- Enter the user name assigned to the caller/calling device in the **User ID** field. SETU ATA211G will use this User ID to match the digest credentials sent by the caller/calling devices when challenged.

Make sure the User ID you enter here and the User ID assigned at the calling end are the same. The User ID can be up to 40 characters long. Default: Blank.

- Enter the password to authenticate the user ID in the **User Password** field. The password may consist of a maximum of 24 characters. Default: Blank.

Make sure the User Password you enter here and the User Password assigned at the calling end are the same.

- Click **Submit** to save the entries.
- Now, enable Digest Authentication on the desired SIP Trunk. See [“SIP Trunks”](#) for instructions.

PIN Authentication

SETU ATA211G supports PIN Authentication for outgoing calls from the FXS port and for incoming calls on the Mobile port and SIP trunks.

You may use PIN Authentication on the Mobile port, the SIP trunks and the FXS port to restrict access to SETU ATA211G to specific callers.

The PIN Authentication feature works on the basis of the PIN Authentication Table, which you must configure.

When you enable PIN Authentication on the Mobile port or the SIP trunk, for each incoming call,

- ATA211G will answer the call and play Prompt tone to the caller.
- the caller must dial the PIN.
- ATA211G matches the PIN dialed by the caller with the PIN Authentication Table.
- If a match is found, the caller is authenticated. The caller can now dial the number.
- If no match is found ATA211G will play error tone, followed by Prompt tone. The caller is allowed two more attempts to dial the PIN. If PIN Authentication fails again ATA will disconnect the call.



This feature will work on the Mobile port, only if Incoming Call Routing option is selected as:

- *Answer the call, collect number & route as per incoming call route selection.*
OR
- *Route to FXS. If not answered within ring timer, answer the call and route as per incoming call route selection.*

This feature will work on the SIP trunk, only if Incoming Call Routing option is selected as:

- *Answer the call, collect number & route to FXS.*
OR
- *Answer the call, collect number & route to Mobile.*

Configuring PIN Authentication

To use this feature, make sure you have enabled **PIN Authentication** on the desired port type: FXS, Mobile, SIP.

- On the Mobile port, enable the **Authenticate Incoming Calls** check box. See [“Mobile Port”](#)
- On the SIP trunk, enable the **Authenticate Incoming Calls** check box. See [“SIP Trunks”](#)
- For the FXS port, enable the **Authenticate Outgoing Call** check box. See [“FXS Port”](#)

Now, configure the PIN Authentication Table.

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand the links.

3. Click **PIN Authentication**.

MATRIX SETU ATA211G

Basic Settings

Advanced Settings

- System
- Called Party Number Table
- Dialed Number Table
- Peer-to-Peer Numbers
- Mobile Emergency Numbers
- Digest Authentication
- PIN Authentication**
- Daylight Saving Time
- Static Routing
- Factory Defaults
- Restart
- Syslog
- PCAP
- Software Upgrade
- Auto-Configuration

Supplementary Services

Status

PIN Authentication

Index	PIN	Index	PIN
001		051	
002		052	
003		053	
004		054	
005		055	
006		056	
007		057	
008		058	
009		059	
010		060	
011		061	
012		062	
013		063	

The PIN Authentication Table page opens. You can configure upto 100 entries in this table. This Table is common for authenticating callers on all port types: FXS, Mobile and SIP.

4. Enter the **PIN** to be authenticated against each Index in the Table. The PIN can have a maximum of 4 digits. The characters allowed are: 0 to 9, *, #
Default: Blank.
5. Click **Submit** to save the entries.

Daylight Saving Time

Daylight Saving Time (DST) is the practice of advancing clocks so that afternoons have more daylight and mornings have less. Typically clocks are adjusted forward one hour near the start of Spring and are adjusted backward in Autumn. Many countries of the world⁷ use it, though the start and end dates of DST vary with location and year.

SETU ATA211G supports Daylight Saving Time adjustment to set the the Date and Time⁸ of SETU ATA211G are set forward and backward according to the DST convention followed in your country.

You can set DST by: **Day and Month** or **Date and Month**.



When ATA211G is set to default, your DST settings will remain unchanged.

Configuring DST

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on left pane to expand the links.
3. Click the **Daylight Saving Time** link.

The screenshot shows the 'MATRIX SETU ATA211G' web interface. On the left is a navigation menu with 'Basic Settings', 'Advanced Settings' (expanded), and 'Supplementary Services'. Under 'Advanced Settings', 'Daylight Saving Time' is selected. The main panel is titled 'Daylight Saving Time Adjustments'. It contains a 'Daylight Saving Time' checkbox (unchecked), a 'Time Offset' input field set to '60 min', and a 'Type' dropdown set to 'Day-Month Wise'. Below these is a table for DST start and end dates:

	Ordinal	Day	Month	Time	
				Hours	Minutes
DST Start	1st	Sunday	March	00	00
DST End	1st	Sunday	September	00	00

At the bottom of the panel are 'Submit' and 'Default' buttons.

4. Enable the **Daylight Saving Time** check box. Default: Disabled.
5. Enter the **Time Offset**. This is the time which the system will consider to forward the clock at the start of DST and to set the clock back when DST ends. Default: 60 minutes.

7. In most countries in Asia and Africa, and in certain countries of South America, DST is not observed.

8. SETU ATA211G sets its Date and Time according to the **Time Zone** you selected, and synchronizes the time according to the **NTP Server** you selected. See "[System Parameters](#)".

6. Select the desired DST **Type** as:

- **Day-Month Wise:** Select this option if the DST in your country starts and ends on a particular day of the month. For example, if DST starts on the Second Sunday of March and ends on the First Sunday of October.

OR

- **Date-Month Wise:** Select this option if the DST in your country starts and ends on a particular date of the month. For example, if DST starts on October 12 and ends on March 15.

Default: Day-Month Wise.

7. If you selected the **Day-Month Wise** option, configure the Start and End time for DST.

DST Start

- Select the **Ordinal** day of the month when DST begins: 1st, 2nd, 3rd, 4th or 5th.
- Select the **Day** of the month when DST begins: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
- Select the **Month** when DST begins: January to December.
- Set the **Time** when you want DST to begin. The time mode is in 24 hours format.

Default: 1st, Sunday, March, Time 00 hours and 00 minutes.

DST End

- Select the **Ordinal** day of the month when DST ends: 1st, 2nd, 3rd, 4th or 5th.
- Select the **Day** of the month when DST ends: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
- Select the **Month** when DST ends: January to December.
- Set the **Time** when you want DST to end. The time mode is in 24 hours format.



When the DST of a particular country starts or ends on the Last Sunday or any other day, e.g. the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

8. If you select **Date-Month Wise** option, configure the following parameters:

DST Start

- Select the **Month** when DST begins: January to December.
- Select the **Date** on which DST begins: 1 to 31.
- Set the **Time** when DST begins. The time mode is in 24 hours format.

DST End

- Select the **Month** when DST ends: January to December.
- Select the **Date** on which DST ends: 1 to 31.
- Set the **Time** when DST ends. The time mode is in 24 hours format.

9. Click **Submit** to save your DST settings.

Example: If you are installing SETU ATA211G in a country in the European Union, as per the European Summer Time, the DST would start on the Last Sunday in March and end on the Last Sunday in October each year. Clocks

are advanced by one hour at 01:00 hours GMT at the start of DST and set back by one hour at 01:00 hours GMT when DST ends.

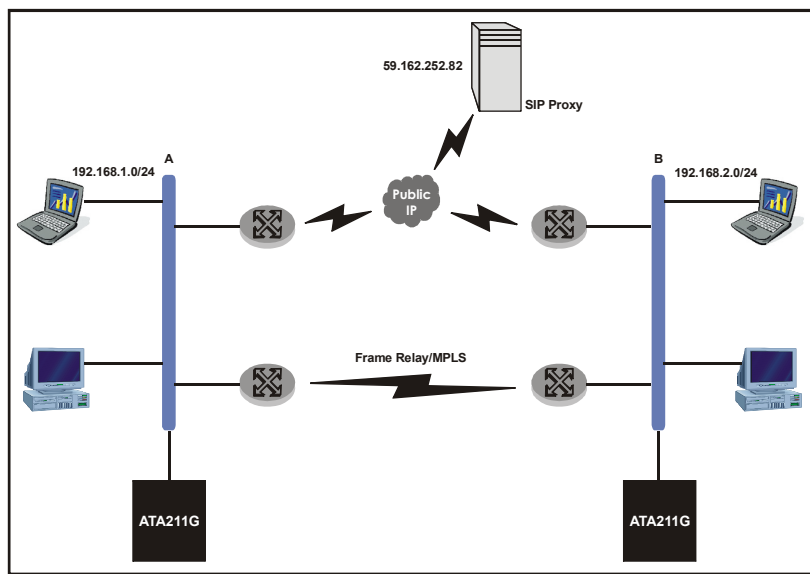
1. Select the **Enable** check box to enable **Daylight Saving Time**.
2. Set the **Time Offset** as 60 minutes.
3. Select the **Day-Month Wise** option as **DST Type**.
4. Configure the **DST Start** as follows:
 - Select **5th** as the **Ordinal**.
 - Select **Sunday** as the **Day**.
 - Select **March** as the **Month**.
 - Set Time to 01 Hours and 00 Minutes.
5. Now, go to the **DST End** option, and configure as follows.
 - Select **5th** as the **Ordinal**.
 - Select **Sunday** as the **Day**.
 - Select **October** as the Month.
 - Set Time to 01 Hours and 00 Minutes.
6. Click **Submit** to save DST settings.

Static Routing

Static Routing Table is required when you have more than one router (Gateway) in your network and you want SETU ATA211G to send packets to multiple routers/gateways for different types of calls.

To illustrate this with an example, two Local Area Networks, Network A and Network B, are connected through Frame Relay/ MPLS network to give access to local resources and also to make Peer-to-Peer calls.

SETU ATA211G connected in a Frame Relay/ MPLS network



- SETU ATA211G is connected at both sites behind a router.
- These sites are also connected to public IP network for the following reasons:
 - To give internet access to local hosts.
 - To access DID service provided by ITSPs to make PSTN/ GSM calls over IP network.
- Network A and Network B are in different subnets.

The Static Routing Table makes it possible to route different types of outgoing calls - Peer to Peer or Proxy - made to different subnets through different Gateways.

The Static Routing Table defines the appropriate Gateway Address (or Router's LAN Address) where the IP packets are to be sent.

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent.
- The Subnet Mask to be applied on the final destination address.
- The Gateway Address where the IP packets are to be sent.

When SETU ATA211G sends packets, if the final destination IP Address and SETU ATA211G are not in the same Subnet, the system will check the Static Routing Table.

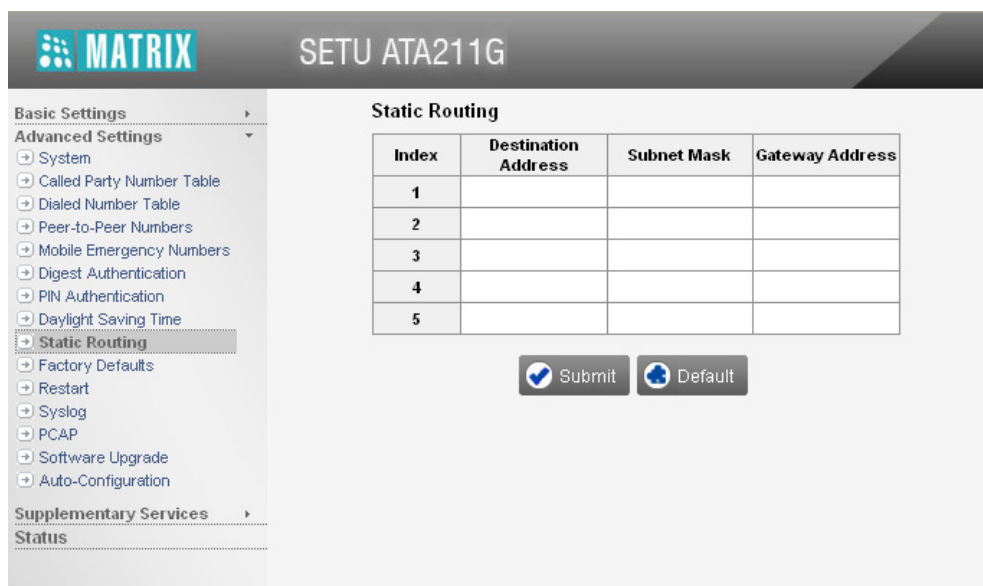
If a perfect match is found, SETU ATA211G will start sending the IP packets to the corresponding Gateway Address configured in the table.

If no match is found, SETU ATA211G will send the IP Packets to the default Gateway Address you configured in the “[Network Parameters](#)” page.

Configuring Static Routing Table

The Static Routing Table must be configured at each location where SETU ATA211G is installed.

1. Log in as **Admin**.
2. Click the **Advanced Settings** link to expand.
3. Click **Static Routing**.
The Static Routing Table opens.



The screenshot shows the SETU ATA211G web interface. The sidebar on the left contains the following links: Basic Settings, Advanced Settings (expanded), System, Called Party Number Table, Dialed Number Table, Peer-to-Peer Numbers, Mobile Emergency Numbers, Digest Authentication, PIN Authentication, Daylight Saving Time, Static Routing (selected), Factory Defaults, Restart, Syslog, PCAP, Software Upgrade, Auto-Configuration, Supplementary Services, and Status. The main content area is titled 'Static Routing' and contains a table with 5 rows and 4 columns: Index, Destination Address, Subnet Mask, and Gateway Address. Below the table are two buttons: 'Submit' and 'Default'.

Index	Destination Address	Subnet Mask	Gateway Address
1			
2			
3			
4			
5			

The Static Routing Table allows you to configure up to 5 entries. Each entry is stored against an Index number.

For each entry, you must configure the following fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- **Subnet Mask:** This is the mask to be applied on destination address.
- **Gateway Address:** This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router (to which SETU ATA211G is connected).

The Gateway Address must be in the same subnet as SETU ATA211G.

4. Click **Submit** to save your entries.

For example, the Static Routing Table of the SETU ATA211G at Location A, will be as shown below.

Index	Destination Address	Subnet Mask	Gateway Address
1	192.168.2.0	255.255.255.0	192.168.1.1
:			
5			

Here, the Destination Address 192.168.2.0 specifies the network address of Location B.

The Subnet Mask is the mask to be applied on Destination address.

The Gateway Address 192.168.1.1 specifies the LAN address of the Router used to connect SETU ATA211G at Location A with SETU ATA211G at Location B.

Now, the IP address of LAN interface of the Router which connects Location A to the Public Internet should be configured as the 'Default Gateway' in the Network parameters.

With the Static Routing Table configured as shown above, packets sent by SETU ATA211G to 192.168.2.0/24 will be routed to the router which connects Location A to location B. Whereas, all packets sent to an address other than 192.168.2.0/24 will be routed to the Default Gateway.

Reinstate Factory Defaults

SETU ATA211G enables you to restore factory settings to all the programmable parameters using the option **Factory Defaults**.

You can reinstate Factory Defaults by logging in as Admin or User. The factory set values will be assigned to all parameters depending on the type of login, Admin or User.

When you log in as **Admin** to reinstate factory defaults, all parameters will be assigned default values except the following:

- Time Parameters (see ["Region"](#))
- Daylight Saving Time (DST) Parameters (see ["Daylight Saving Time"](#)).
- CPTG and Ring (see ["Region"](#)).

When you log in as **User** to reinstate factory defaults, default values will be assigned to the parameters listed below:

- Supplementary Features
- Speed Dialing
- Network
- User Password
- Syslog

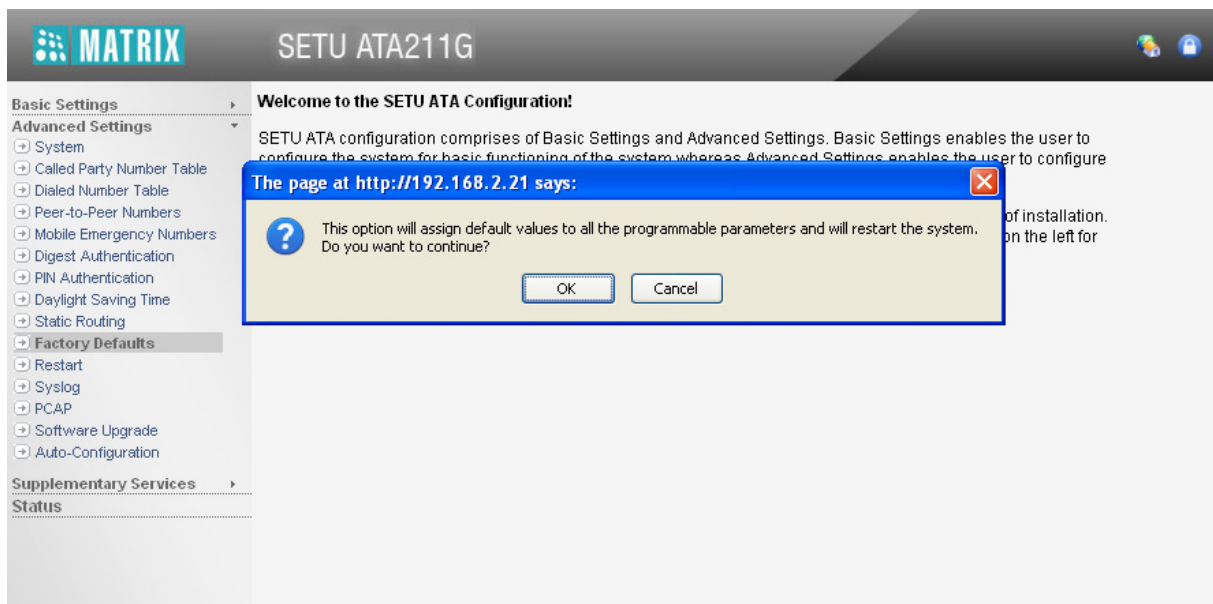
From the User Login, the Admin Password will be set to its default value, 1234, if you have enabled the option 'Default Admin password when 'User' defaults the ATA?' in the System Parameters. See .

Restoring Factory Default

Using Jeeves

1. You can login either as **Admin** or **User** with your Password.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Factory Defaults**.

An alert message will be displayed, **This option will assign default values to all the programmable parameters and will Restart. Do you want to continue?**



4. Click **OK**.

The system will restart.

To restore Factory Defaults from User mode,

1. Log in as **User**.
2. Click **Factory Defaults** link under **User Settings**.
3. Click **OK** on the alert message: **This option will assign default values to all the programmable parameters and will Restart. Do you want to continue?**

The system will restart.

Using Commands

1. Lift the handset of the phone.
2. To log in as Admin, dial the programming access code **#19** followed by the Admin Password (**default is 1234**).

OR

3. To log in as User, dial the access code **#18** followed by User Password (**default is 1234**).
4. You will get programming tone.
5. Dial **21-Reverse Admin/User Password-#***
For example: **21-4321-#*** (the reverse of the default password 1234).

In case you have forgotten the password, set the User / Admin Password to default. See [“Passwords”](#) for instructions.

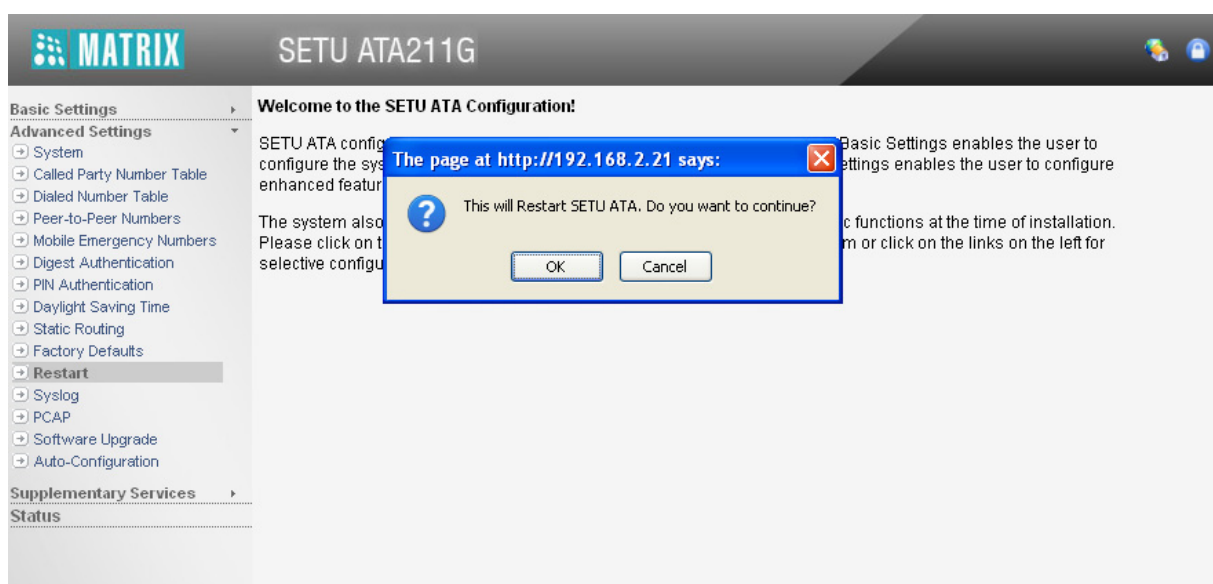
Restart SETU ATA211G

If you need to restart SETU ATA211G, you may do it using Jeeves, instead of switching OFF and switching ON the system again.

When you restart the system, all active calls will be disconnected and the ports in use will be released. The system configuration however, will not be affected.

To restart the system using Jeeves,

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Restart**.



4. An alert message will appear, '**This will Restart System. Do you want to continue?**'
5. Click **OK** button to restart the system.

Syslog

Syslogs (debugs) are logs of actions and events that take place on any computer system. These logs are useful for troubleshooting and system security.

SETU ATA211G supports Syslog⁹ Client for debugging. Syslog Client enables the system to send debug messages in syslog format to the remote 'Syslog Server' on IP network. You can view the system debug messages on the remote server.

Each debug message includes the MAC Address of SETU ATA211G which is sending debug messages in 'syslog server'.

ATA211G supports following different Syslog levels:

- APPL
- CONFIG
- SYS
- SIP
- CALL
- DSP
- REG
- SNTP
- STACK
- NAT
- AUTO CONFIG
- Mobile

You can select any of these syslog levels, and the Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of **Call** is required, you can select this option, and disable all others.

To be able to use this feature, you must enable **Syslog**, configure the Syslog Server Address and define the Server Port on which Syslog will listen for debug messages.

Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

To debug the System from the Admin mode,

1. Log in as **Admin** with your Password.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **Syslog**.

9. Syslog is one of the protocols used extensively for sending debug messages, and is defined in RFC 3164.

MATRIX SETU ATA211G

Basic Settings

Advanced Settings

- System
- Called Party Number Table
- Dialed Number Table
- Peer-to-Peer Numbers
- Mobile Emergency Numbers
- Digest Authentication
- PIN Authentication
- Daylight Saving Time
- Static Routing
- Factory Defaults
- Restart
- Syslog**
- PCAP
- Software Upgrade
- Auto-Configuration

Supplementary Services

Status

Syslog

Syslog ☐ Enable

Syslog Server Address:Port : 514

Syslog Levels

☒ APPL	☒ CALL	☒ STACK
☒ CONFIG	☒ DSP	☒ NAT
☒ SYS	☒ REG	☒ AUTO CONFIG
☒ SIP	☒ SNTP	☒ Mobile

4. Configure the following parameters:

- **Syslog:** Select the check box to enable. By default, disabled.

When enabled, the system will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the system.

- **Syslog Server Address : Port:** Enter the IP Address of the Syslog Server. A maximum of 15 digits, including 0-9 and '.' (dot) are allowed.

Enter the Listening Port of the Syslog Server from 1024-65535: 514. By default the Syslog port is 514.

- **Syslog Levels:** Select the desired debug level by selecting the check box. The system will send debug messages only for the level you have selected. By default, all debug levels are selected. Clear the check box to disable the debug level you do not want.

5. Click the **Submit** button to save the entries.

To debug the System from User mode,

1. Log in as **User**.
2. Click the **Syslog** link, under **User Settings** on the left pane.
3. To configure Syslog settings, refer the steps 4 and 5 described for the Admin mode.

PCAP

PCAP or packet capture consists of intercepting and logging the traffic passing over a digital network or a part of a network. PCAP intercepts each packet in the data streams that flow across the network, and can decode and analyze its contents.

PCAP can be used, among others, to monitor the network, analyze network problems, debug client/server communications, debug network protocol implementations.

SETU ATA211G supports PCAP Trace, which you can use to detect and diagnose network related problems, for example, when the SIP account is not getting registered, or any SIP related feature is not functioning.

Packets traveling over a network are captured and saved in the system. You can save these trace files (packets captured by the system) on a computer and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 1 MB of packets can be captured and stored in the system.

SETU ATA211G also supports Filters and Promiscuous mode for capturing packets, which you can use to specify the types of data packets to be captured.

To use PCAP Trace from Admin mode,

1. Log in as **Admin**.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **PCAP**.

PCAP Trace

Filter Setting:

Promiscuous Mode: ☒ Enable

Packets Captured:

Total Bytes:

Status: m pcap : off for 0

Examples of Filter Setting

Filter Type	Filter Setting	Comment
src port <i>port number</i>	src port 5060	Capture packets if the packet has a source port value of 5060.
dst port <i>port number</i>	dst port 80	Capture packets if the packet has a destination port value of 80.
port <i>port number</i>	port 5060	Capture packets if the packet has either source or destination port value of 5060.
src host <i>ip address</i>	src host 192.168.1.176	Capture packets if the source IP address is 192.168.1.176
dst host <i>ip address</i>	dst host 192.168.1.176	Capture packets if the destination IP address is 192.168.1.176.
host <i>ip address</i>	host 192.168.1.176	Capture packets if either source or destination IP address is 192.168.1.176

Start

4. Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter should be maximum 60 characters in length. By default, this field is blank. So, all packets will be captured.

Refer to the following examples to know how to set the Filters.

Examples of Filter settings:

- To capture packets of source port 5060, set the filter as **src port 5060**
- To capture packets of destination port 80, set the filter as **dst port 80**
- To capture packets transmitted from IP address 192.168.1.176, set the filter as **src host 192.168.1.176**
- To capture packets received from IP address 192.168.1.176, set the filter as **dst host 192.168.1.176**



It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.

5. You may select the check box to enable **Promiscuous Mode** if you want. Default: Disabled.

When you enable Promiscuous Mode, the SETU ATA211G will capture all network traffic. However, this will work only in a non-switched environment.

When Promiscuous Mode is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through the SETU ATA211G will be picked up by the PCAP Trace.



'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

6. Click the **Start** button to begin the capturing of the packets.
7. Click the **Stop** button to stop packet capture.

OR

Wait for the system to stop packet capturing. The system stops packet capturing once the maximum allotted memory of 1 MB (RAM) is utilized.

The **Status** field displays the current activity of packet capturing.

Number of Packets and bytes captured as per the filter setting will be displayed as **Packets Captured** and **Total Bytes**.



Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.

8. When the packet capturing is stopped (by you or the system), click the **Save Trace File** button to save the files on your computer or on another computer.

A dialog box open. You can select the path for saving the trace file.



The current packets captured will not be deleted after you have saved the trace file. The current packets will be deleted when you start the PCAP capture again.

9. You may log out of Jeeves.
10. Now, you can open the trace files using Wireshark/Ethereal or any other similar software which supports opening of trace files.

To use PCAP Trace from User mode,

1. Log in as **User**.
2. Click the **PCAP** link, under **User Settings** on the left pane.

Matrix SETU ATA211G

User Settings

- Supplementary Features
- Speed Dialing
- Network
- User Password
- Syslog
- Factory Defaults
- PCAP**

Status

PCAP Trace

Filter Setting

Promiscuous Mode ☐ Enable

Packets Captured

Total Bytes

Status

Examples of Filter Setting

Filter Type	Filter Setting	Comment
src port <i>port number</i>	src port 5060	Capture packets if the packet has a source port value of 5060.
dst port <i>port number</i>	dst port 80	Capture packets if the packet has a destination port value of 80.
port <i>port number</i>	port 5060	Capture packets if the packet has either source or destination port value of 5060.
src host <i>ip address</i>	src host 192.168.1.176	Capture packets if the source IP address is 192.168.1.176
dst host <i>ip address</i>	dst host 192.168.1.176	Capture packets if the destination IP address is 192.168.1.176.
host <i>ip address</i>	host 192.168.1.176	Capture packets if either source or destination IP address is 192.168.1.176

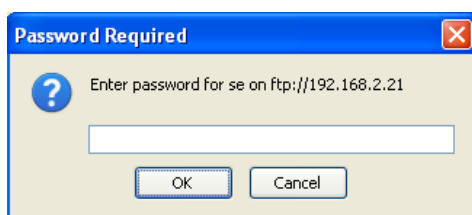
3. To configure PCAP settings, refer the steps 4 to 10 described for the **Admin** mode.

Software Upgradation

SETU ATA211G provides you the facility to upgrade the system software at the click of a button. SETU ATA211G supports an embedded FTP¹⁰ server which can be used for Uploading and Downloading System files, configuration files, drivers etc.

To update configuration settings and firmware,

1. Log in as **Admin** with your Password.
2. Click the **Advanced Settings** link on the left pane to expand.
3. Click **System Upgrade**. The following message box will be displayed.



4. In the Password field of the window, enter the Admin Password.
5. The following folders will be displayed:
 - **system**: contains the current firmware/software.
 - **config**: contains the configuration files.
 - **driver**: contains the current driver files.
 - **web**: contains html files.
 - **dhcp**: contains files of DHCP configuration.
 - **ppp**: contains files for PPPoE configuration.
6. Click any of these folders to view the files that SETU ATA211G is currently using. You may either delete the existing files or copy the existing files to another location (as backup).
7. Copy the new configuration files from their location and paste these new files in these folders.

Wait for the file transfer to complete.

8. Restart the system after uploading the files. The new configuration/driver/firmware will be applicable only after the system restarts.



Before you upgrade software, you can save the existing application/configuration/driver/ web pages folders and files as back-up for retrieval, later. For example: if you want to retain the existing Configuration folder for future record, you can save the folder on your computer. When required, you can copy and paste the same folder back on the FTP server of SETU ATA211G.

¹⁰.FTP or **File Transfer Protocol** is a standard Internet Protocol that is used to exchange files between computers on the IP network.

Auto Configuration

You can use Auto-Configuration to have the system automatically download the configuration files stored at a central location: TFTP/HTTP/HTTPS Server.

This feature is useful for ITSPs and where multiple SETU ATA211G are connected. ITSPs can store the configuration files of each SETU ATA211G that they provide to their customers on the Auto Configuration Server (ACS). The configuration files of each ATA are to be stored in a folder named by its MAC Address.

When a customer connects to the ITSP network, the SETU ATA211G provided to the customer will automatically download its configuration files stored in its MAC Address folder on the ACS. The customer can thus start using the SETU ATA211G without the intervention or assistance of a technician, making it a 'plug-and-play' device.

To use this feature, end users must obtain from the ITSP or System Administrator the following information:

1. IP address of the Auto Configuration Server.
2. Path of the Folder (containing the configuration files) on the Auto Configuration Server.
3. The protocol to be used: TFTP, HTTP, HTTPS.

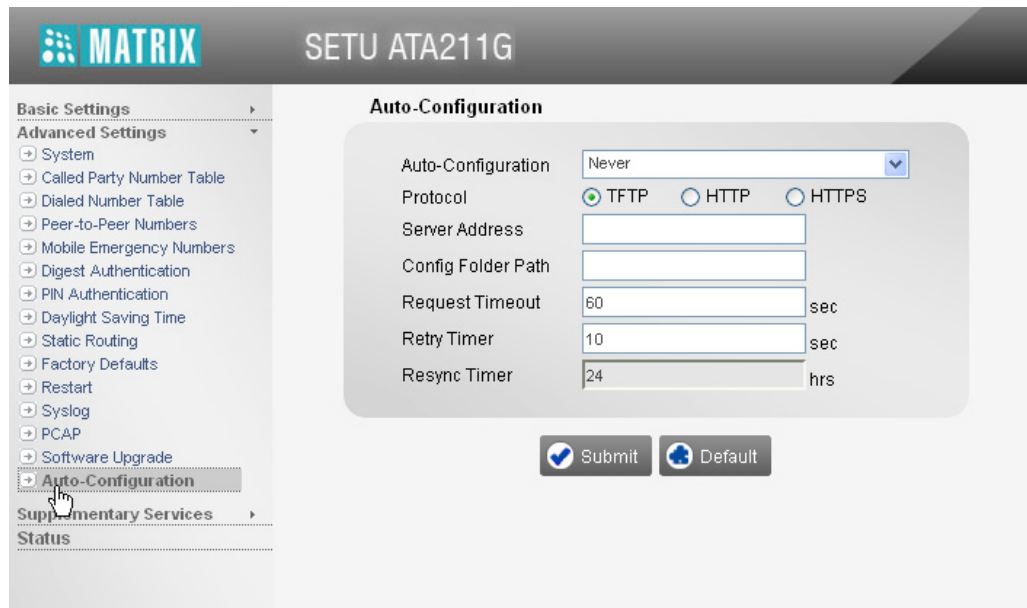
When your ACS uses HTTP, HTTPS protocol for Auto-Configuration, make sure:

- The configuration files of each ATA are stored in a folder by its MAC Address.
- Store the index.html or startup.html or any other default file name used by your server in the MAC Address folder of each ATA.
- This file should include the names of the configuration files of ATA in the same format as provided to you on the CD-ROM supplied to you with your ATA. See the folder name **Auto Configuration Sample File** on the CD.

When ATA sends GET request to perform Auto-Configuration, server should return index.html or startup.html or other file, which includes the configuration file names to be supplied to ATA. Once ATA receives the file (index.html or startup.html or other) with configuration file names, ATA will send separate requests for each configuration file.

Using Auto Configuration:

1. Log in as **Admin** with your Password.
2. Click the link **Advanced Settings** on the left pane to expand.
3. Click **Auto Configuration** and configure the following parameter.



4. **Auto Configuration:** Select the desired method to be used to fetch configuration files. Default: Never.

- **Never:** Select this option if you do not want Auto-Configuration process to be done during power ON.
- **At Each Power ON:** Select this option if you want SETU ATA211G to perform Auto-Configuration at each power ON.
- **At Next Power ON:** Select this option if you want SETU ATA211G to perform Auto-Configuration only during the next power ON. Once Auto-Configuration is done successfully, the option gets changed to **Never**.
- **Resync Periodically:** Select this option when SETU ATA211G does not need to auto configure during power ON, but needs to resynchronize periodically with the ACS to check for updates in the configuration. If you select this option, Auto-Configuration will be performed every time the Resync Timer ends.
- **At Each Power ON & Resync Periodically:** Select this option when SETU ATA211G needs to auto configure during power ON and also to resynchronize periodically with the ACS to check for updates in the configuration.
- **At Next Power ON & Resync Periodically:** Select this option when SETU ATA211G needs to auto configure only during the next power ON but resynchronize periodically with the ACS to check for updates in the configuration. Once Auto-Configuration is done successfully at next power on, the option gets changed to **Resync Periodically** option.

5. **Protocol:** Select the protocol used by the Auto-Configuration Server for file transfer. The system generates file transfer request to the configuration server according to the protocol you selected. Default: TFTP

- **TFTP:** Select the radio button, if you want this protocol for the process of Auto-Configuration. TFTP is preferred when Auto Configuration is to be done in Local Area Network (LAN).
- **HTTP:** Select the radio button, if you want this protocol for the process of Auto-Configuration. HTTP is preferred when Auto-Configuration is to be done in Wide Area Network (WAN).

- **HTTPS:** Select the radio button, if you want this protocol for the process of Auto-Configuration. HTTPS is preferred when security is the concern during Auto-Configuration.

6. **Server Address:** Specify the Auto-Configuration Server Address. The Auto-Configuration Server Address resembles the IP Address/Domain of ACS where all configuration files of SETU ATA211G are stored. The server address can be configured using DHCP.

If the server address is to be configured manually, enter TFTP/HTTP/HTTPS server's IP address from where ATA should download the configuration files. Default: Blank.

If you want the system to get the Server Address from the DHCP server, keep this field blank. Make sure that you also set the 'Connection Type' on the Network page to 'DHCP'.

7. **Config Folder Path:** Specify the path of the folder (provided by your ITSP / System Administrator) on the TFTP/HTTP/HTTPS server from where the configuration files are to be downloaded.

When SETU ATA211G gets auto configured, it will send the request for the configuration files to the selected TFTP/HTTP/HTTPS server and restore the files from the folder as per this path.

Default: Blank.

8. **Request Timeout:** Request Timeout is used when SETU ATA211G tries to connect to the Auto-Configuration Server for TCP/TLS binding. This timer specifies for how long SETU ATA211G should wait for successful TCP/TLS binding.
Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

9. **Retry Timer:** This timer is used if the connection is not established with Auto-Configuration Server before the expiry of the Request Timeout. Then, in such case, SETU ATA211G will try to reconnect automatically after this time interval.

Enter the required time interval in seconds. The range of Retry Timer is 001-999 seconds. Default: 10 seconds.

10. **Resync Timer:** Resync Timer is the time after expiry of which SETU ATA211G will send the resync request to check if there is any change in the configuration files. This timer is applicable only when **Resync Periodically** or **At Each Power ON & Resync Periodically** or **At Next Power ON & Resync Periodically** options of Auto-Configuration are selected.

Enter the required resync timer in hours. The range of Resync Timer is 001-999 hours. Default: 24 hours.

11. Click **Submit** to save.

Call Hold

Call Hold enables you to put an on-going conversation on hold, and call another person or receive a call from another person. You can retrieve the call you put on hold, after the conversation with the other party has ended. You can also retrieve the call you put on hold in the middle of the conversation with the other party.

Call Hold is also used in other features like Making Second Call, Call Waiting, Call Transfer- Blind, Call Transfer- Attended, Call Toggle and 3-Party Conference.

Configuring Call Hold



To use this feature, make sure that the Call Hold feature is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.

Using Call Hold

- You are in speech with party A.
- To put your call with A on hold, dial **Flash**.
- A is put on hold. You get feature tone for 7 seconds, followed by error tone.
- To retrieve the call you put on hold, dial **Flash** again during feature tone or during error tone.
- You will be in speech with A.



- If you go ON-Hook during the feature tone, your call with party A will be disconnected.
- If you go ON-Hook during error tone, you will get ring back. If you go OFF-Hook when your phone rings, you will get connected with party A again.

Making a Second Call

You can make a second call, by putting the current call on hold.



*To use this feature, make sure that the Call Hold feature is enabled in the **Class of Service** of the FXS port. See “[FXS Port](#)” topic.*

Making a Second Call

- You are in speech with party A.
- Dial **Flash** to put party A on Hold.
- You get feature tone.
- Dial the number of the desired party B.
- When party B answers the call, you are in speech with party B.



Making a second call feature can also be used with other features such as Call Transfer-Attended, Call Toggle and 3-Party Conference.

Call Transfer

Using Call Transfer you can connect the person you are in speech with, with another party.

SETU ATA211G offers two types of Call Transfer:

- **Call Transfer - Blind:** You can transfer the call to the desired party, without informing the party of the transfer.
- **Call Transfer - Attended:** You can transfer the call to the desired party after consulting the party and or obtaining their consent for transfer.

When you transfer a call, SETU ATA211G will wait for the notification from the destination number to which you have transferred the call, for the duration of the 'Transfer Notification Timer'. This timer is set to 60 seconds by default. It can be set to the desired value.

You will be notified about the success or failure of the transfer attempt during this Timer. You will get a confirmation tone, when the call transfer is successful.



*To use this feature, make sure that the Call Transfer feature is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.*

*You may change the default value of the Transfer Notification Timer to the desired duration, if required, from the [“”](#) topic under **Advanced Settings**.*

Using Call Transfer

Call Transfer-Blind

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial **Flash-#6**.
- Party A is put on hold, and you will hear dial tone.
- Dial the number of party B.
- On successful transfer, you will get confirmation tone.
- Party A will get connected with party B.
- Replace the handset of your phone.



When you transfer a call, remain Off-hook, i.e. do not replace your handset, after dialing the transfer destination's number, until you know the result of the call transfer. One of the following results may occur:

- *If the transfer is successful, you will get confirmation tone for 3 seconds, followed by error tone.*
- *If the transfer has failed, you will get error tone for 2 seconds and you will be connected to the transferee (party A, in this case).*
- *If the transfer has timed out, i.e. the Notification Timer has expired and no notification has been received from the transfer target (party B, in this case) you will get continuous error tone.*

Call Transfer-Attended

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial **Flash**.
- Party A is put on hold, and you will hear dial tone.

- Dial the number of party B.
- When party B answers the call, you will be in speech with party B.
- You may now transfer the call if party B consents.
- Dial **Flash-#7** to transfer the call to party B.
- On successful transfer, you will get confirmation tone, followed by dial tone.
- Party A will get connected with party B.
- Replace the handset of your phone.



To know the call transfer result, remain Off-hook after dialing the transfer target's number (party B).

- *If the transfer is successful, you will get a confirmation tone for 3 seconds followed by error tone.*
- *If the transfer has failed, you will get error tone for two seconds and will be connected to the transfer target (party B), if available. If the transfer target has disconnected the call, you will be connected to the transferee.*
- *If the transfer has timed out, you will get continuous error tone.*

Call Toggle

Call Toggle allows you to have two simultaneous telephone conversations, talking to two persons alternately.

The parties for Call Toggle can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.



*To use this feature, make sure that Call Toggle is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.*

Using Call Toggle

- You are in speech with A and you want to talk to B.
- Dial **Flash**, A is put on hold. You get feature tone.
- Dial the number of B. When B answers the call, you are in speech with B.
- To talk to A, dial **Flash-#4**.
- You are in speech with A and B is put on hold.
- To talk to B, dial **Flash-#4** again.
- You are in speech with B. A is put on hold.
- This way, you can talk alternately to A and B, by dialing **Flash-#4** again.



When toggling between calls, you can disconnect the call you are currently in speech with by going On-hook.

Call Waiting

When your phone is busy, the Call Waiting feature notifies you about another incoming call in the form of beeps.

The Call Waiting feature of SETU ATA211G allows you to:

- reject the waiting call.
- hold the current call and attend to the waiting call.
- disconnect the current call and attend to the waiting call.

Configuring Call Waiting



To use this feature, make sure that the Call Waiting feature is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.

Using Jeeves

To configure Call Waiting from the Admin mode:

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click **Supplementary Features**.

The screenshot shows the 'Supplementary Features' configuration page for the SETU ATA211G. On the left, a navigation menu lists 'Basic Settings', 'Advanced Settings', 'Supplementary Services' (expanded), 'Supplementary Features' (selected), 'Speed Dialing', and 'Status'. The main content area has a title 'Supplementary Features' and contains several checkboxes: 'Call Forward - Unconditional', 'Call Forward - Busy', 'Call Forward - No Reply', 'Enable Call Waiting' (checked and highlighted with a red box), and 'Enable Do-Not-Disturb'. There is also a 'Timer' field set to '45' with a unit of 'sec'. At the bottom, there are 'Submit' and 'Default' buttons.

4. Keep the **Enable Call Waiting** check box enabled.
5. Click **Submit** to save the settings.
6. You may log out.

To configure Call Waiting from the User mode:

1. Log in as **User**.
2. Click the **User Settings** link.
3. Click the **Supplementary Features** link.

The screenshot shows the 'MATRIX SETU ATA211G' web interface. On the left is a 'User Settings' sidebar with a tree view containing 'Supplementary Features' (selected), 'Speed Dialing', 'Network', 'User Password', 'Syslog', 'Factory Defaults', and 'PCAP'. Below this is a 'Status' section. The main content area is titled 'Supplementary Features' and contains several checkboxes: 'Call Forward - Unconditional', 'Call Forward - Busy', 'Call Forward - No Reply', 'Enable Call Waiting' (which is checked and highlighted with a red rectangle), and 'Enable Do-Not-Disturb'. To the right of these checkboxes are input fields; the 'Timer' field is set to '45' with a 'sec' label. At the bottom of the main area are two buttons: 'Submit' (with a checkmark icon) and 'Default' (with a reset icon).

4. Keep the **Enable Call Waiting** check box enabled.
5. Click **Submit** to save the settings.
6. You may log out.

Using Commands

- Lift handset.
- Dial **#19-Admin Password** (default: 1234).
OR
- Dial **#18-User Password** (default: 1234).
- Dial **72-1-#*** to enable.
- Dial **72-0-#*** to disable.
- Replace handset.

Using Call Waiting

When call waiting is enabled and you are busy on a call with party A,

- You will hear beeps indicating another incoming call.
 - You may choose to do any of the following:
 - *Reject the call*, by dialing **Flash-#3**. The beeps will stop, you will remain in speech with party A.
- OR
- put party A on hold, by dialing **Flash-#4**. The beeps will stop, you will be in speech with party B.

OR

- disconnect your call with party A, by dialing **Flash-#5** and answer the waiting call. The beeps will stop, you will be in speech with party B. Party A will be disconnected.

Conference

Three-party Conference, also referred to as Three-Way Calling, is a telephone call, in which you can have two other persons participate in the call.

You can initiate a conference by calling the first person, and then put the first person on hold to call the second person. You can also include another person when you in the middle of speech with a person.

The parties to a conference can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.



- *While initiating the conference, the SETU ATA211G supports the Vocoder G.711 (Mu-law) only.*
- *To use this feature, make sure that the Conference feature is enabled in the **Class of Service** of the FXS port. See “[FXS Port](#)” topic.*

Using Conference

- Lift handset.
- Dial the number of party A.
- When A answers the call, dial **Flash**.
- Party A is put on hold, and you will hear feature tone.
- Now, dial the number of party B.
- When B answers the call and you are in speech with B, dial **Flash-#2**.
- Three-way speech will be established between you, party A and party B.
- Replace the handset at the end of your conversation.
- The conference will be terminated.



*You can convert a 3-Party Conference to Call Toggle by dialing the Call Toggle access code **Flash-#4**. You will be connected to one party and the other party is put on hold.*

Do-Not-Disturb

If you do not want to receive any calls on your phone, you may set the Do-Not-Disturb feature on your phone. When you set DND, all incoming calls on your phone will be rejected, but you can continue to make outgoing calls.

Configuring Do-Not-Disturb



To use this feature, make sure that Do Not Disturb feature is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.

Using Jeeves

To set or cancel DND from Admin mode,

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click **Supplementary Features** link.

The screenshot shows the Matrix SETU ATA211G web interface. On the left is a navigation menu with 'Basic Settings', 'Advanced Settings', 'Supplementary Services' (expanded), 'Supplementary Features' (selected), 'Speed Dialing', and 'Status'. The main area is titled 'Supplementary Features' and contains several checkboxes: 'Call Forward - Unconditional', 'Call Forward - Busy', 'Call Forward - No Reply', 'Enable Call Waiting', and 'Enable Do-Not-Disturb'. The 'Enable Do-Not-Disturb' checkbox is checked and highlighted with a red box. Below these is a 'Timer' field set to '45' seconds. At the bottom are 'Submit' and 'Default' buttons.

4. To set DND, select the **Do-Not-Disturb** check box. Default: Disabled.

To cancel, clear the check box.

5. Click **Submit** to save.

To set or cancel DND from User mode,

1. Log in as **User**.

2. Click the **Supplementary Features** link, under **User Settings** on the left pane.

The screenshot shows the MATRIX SETU ATA211G web interface. On the left, under 'User Settings', the 'Supplementary Features' link is selected. The main content area is titled 'Supplementary Features' and contains the following settings:

- ☐ Call Forward - Unconditional
- ☐ Call Forward - Busy
- ☐ Call Forward - No Reply
- Timer: 45 sec
- ☒ Enable Call Waiting
- ☒ Enable Do-Not-Disturb (highlighted with a red box)

At the bottom of the settings area are two buttons: 'Submit' and 'Default'.

3. To set DND, select the **Do-Not-Disturb** check box. Default: Disabled.

To cancel, clear the check box.

4. Click **Submit** to save.

Using Phone

- Lift handset of your phone.
- Dial **#19-Admin Password** (default: 1234).
OR
- Dial **#18-User Password** (default: 234).
- Dial **61-1-#*** to set.
- Dial **61-0-#*** to cancel.
- Replace handset.

Call Forward

When you are away from your phone¹¹, but would like to answer your calls, you can use the Call Forward feature to have SETU ATA211G forward incoming calls to another number.

SETU ATA211G supports the following Call Forward options, which you can set on your phone:

- **Call Forward-Unconditional:** All incoming calls received on the SIP Trunk are forwarded to the desired destination number, automatically without waiting for a response from your phone.
- **Call Forward- Busy:** All incoming calls received on the SIP Trunk are forwarded to the destination number, when your phone is busy.
- **Call Forward- No Reply:** All incoming calls received on the SIP Trunk are forwarded to the destination number, when you do not answer your phone.

When you set Call Forward-No Reply, SETU ATA211G waits for the duration of the **Call Forward-No Reply Timer** for the phone to answer. This timer is configurable, and is set to 45 seconds as default.

If the phone does not answer before the timer expires, SETU ATA211G considers it as No Reply and forwards it to the desired destination number.

You can set a different destination number for each Call Forward option.

Configuring Call Forward



To use this feature, make sure

- you have selected **Incoming Call Routing** for the SIP Trunk as **Route to FXS**. See [“SIP Trunks”](#) topic.
- Call Forward feature is enabled in the **Class of Service** of the FXS port. See [“FXS Port”](#) topic.

Using Jeeves

To use Call Forward from Admin mode,

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click the **Supplementary Features** link.

¹¹. Connected to the FXS port of SETU ATA211G.

4. Select the check box for the Call Forward option you want to set:
 - **Call Forward-Unconditional**
 - **Call Forward-Busy**
 - **Call Forward-No Reply.**

In the corresponding box enter the desired destination number (up to 40 characters) you want the calls to be forwarded to.

5. If you selected Call Forward- No Reply, you may also change the duration of the no-reply **Timer**, if required. The range of the Call Forward No-Reply Timer is 01 to 99 seconds. Default: 45 seconds.
6. Click **Submit** to save.

To use Call Forward from User mode,

1. Log in as **User**.
2. Click the **Supplementary Features** link under **User Settings** on the left pane.

3. Select the check box for the Call Forward option you want to set:
 - **Call Forward-Unconditional**
 - **Call Forward-Busy**

- **Call Forward-No Reply.**

In the corresponding box enter the desired destination number (up to 40 characters) you want the calls to be forwarded to.

4. If you selected Call Forward- No Reply, you may also change the duration of the no-reply **Timer**, if required. The range of the Call Forward No-Reply Timer is 01 to 99 seconds. Default: 45 seconds.
5. Click **Submit** to save.

Using Phone

Call Forward Un-conditional

To configure destination number for Call Forward-Unconditional:

- Lift the handset of your phone.
- Dial **#19-Admin Password** (default:1234)
OR
- Dial **#18-User Password** (default:1234)
- Dial **51-Destination Number-#***
- Replace handset.

To set/cancel Call Forward-Unconditional:

- Lift the handset of your phone.
- Dial **#19-Admin Password** (default:1234)
OR
- Dial **#18-User Password** (default:1234)
- Dial **52-1-#*** to set.
- Dial **52-0-#*** to cancel.
- Replace handset.

Call Forward-Busy

To configure destination number for Call Forward-Busy:

- Lift the handset of your phone.
- Dial **#19-Admin Password** (default: 1234)
OR
- Dial **#18-User Password** (default: 1234)
- Dial **53-Destination Number-#***.
- Replace handset.

To set/cancel Call Forward- Busy:

- Lift the handset of your phone.
- Dial **#19-Admin Password** (default:1234)
OR
- Dial **#18-User Password** (default:1234)
- Dial **54-1-#*** to set.
- Dial **54-0-#*** to cancel.
- Replace handset.

Call Forward- No Reply

To configure destination number for Call Forward- No Reply:

- Lift the handset of your phone.
- Dial **#19-Admin Password** (default: 1234)
OR
- Dial **#18-User Password** (default: 1234)
- Dial **55-Destination Number-#***.
- Replace handset.

To set the Timer for Call Forward-No Reply:

- Lift handset of your phone.
- Dial **#19-Admin Password** (default: 1234)
OR
- Dial **#18-User Password** (default: 1234)
- Dial **57-Time-#***. Valid range of the Timer is 01-99 seconds.
- Replace handset.

To set/cancel Call Forward-No Reply:

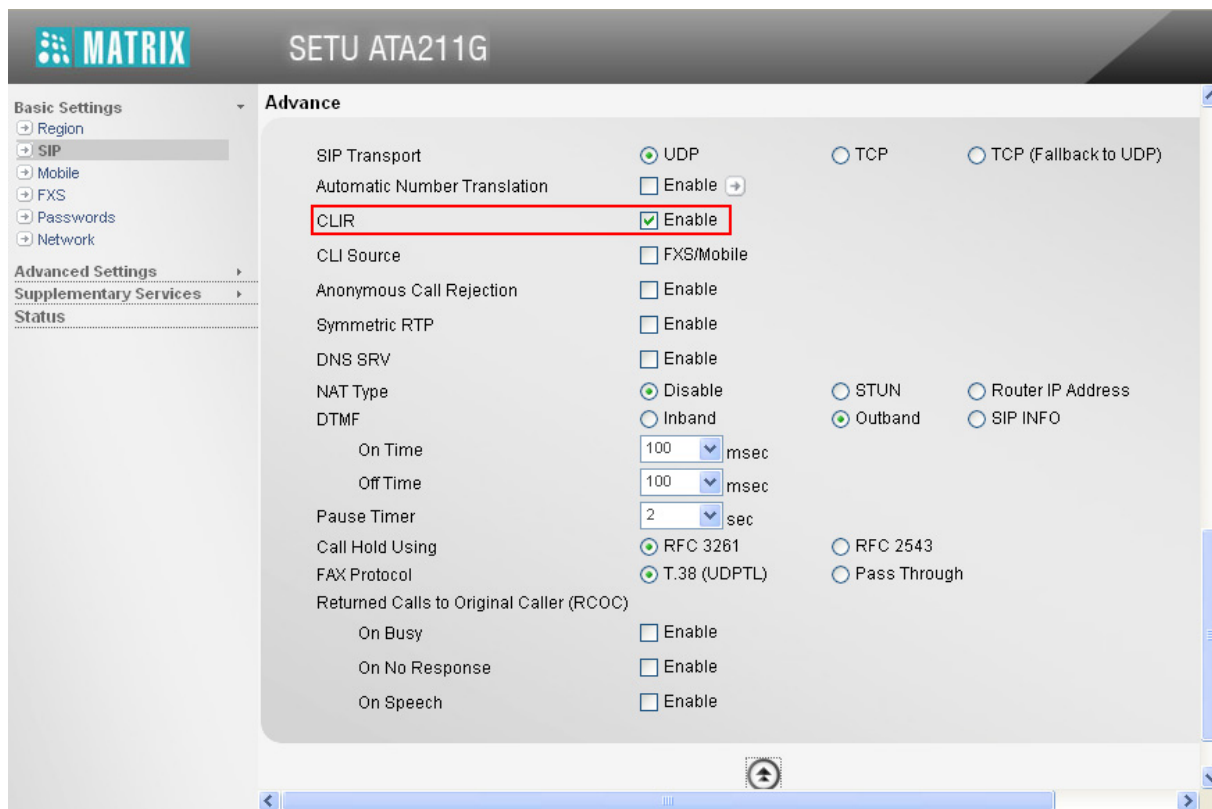
- Lift the handset of your phone.
- Dial **#19-Admin Password** (default: 1234)
OR
- Dial **#18-User Password** (default: 1234)
- Dial **56-1-#*** to set.
- Dial **56-1-#*** to cancel.
- Replace handset.

Caller ID Restriction

If you do not want your caller ID to be displayed to the called party, you can do so using this feature.

Configuring Caller ID Restriction

Using Jeeves



1. Log in as **Admin** with your Password.
2. Click the **Basic Settings** link on the left pane to expand.
3. Click **SIP** to open the SIP trunk parameters page.
4. Select the desired SIP Trunk by clicking the SIP trunk number tab.
5. Click **More** button to view all parameters on the page.
6. Scroll to **Advanced**.
7. Enable **CLIR** to hide your Caller ID. Default: Disabled.
8. Click **Submit** to save.

Using Commands

- Lift handset.
- Dial **#19-Admin Password** (default: 1234).
- Dial **66-SIP Trunk Number -Code-#***

Where,

SIP Trunk Number is

1 for SIP Trunk 1

2 for SIP Trunk 2

3 for SIP Trunk 3

Code is

1 to Enable CLIR

0 to Disable CLIR

- Replace handset.

Selective SIP Trunk Access

Regardless of the SIP Trunk you have selected for 'Outgoing Call Routing', you can make outgoing calls from any SIP Trunk by dialing the Access Code for that trunk.

1. To dial a number from SIP1:
 - Lift the handset of your phone.
 - Dial **#84**.
 - Dial the desired number.
 - Talk, when the remote party answers the call.
2. To dial a number from SIP2:
 - Lift the handset of your phone.
 - Dial **#85**.
 - Dial the desired number.
 - Talk, when the remote party answers the call.
3. To dial a number from SIP3:
 - Lift the handset of your phone.
 - Dial **#86**.
 - Dial the desired number.
 - Talk, when the remote party answers the call.

Speed Dialing

As the name itself suggests, this feature offers you a quick way to dial a number. Using this feature, you can make an outgoing call to a particular number by dialing a short code instead of dialing the entire number string.

Speed Dialing feature works on the basis of the Speed Dialing Table, which you must configure separately for each port.

The Speed Dialing Table consists of *Index Numbers* that serve as short codes. You must assign contact numbers (along with their names) that you want to speed dial against the Index numbers in the table and select the Destination Route through which the number should be dialed out.

Using Speed Dialing from the FXS port,

- Lift the handset.
- Dial the feature access code for Speed Dial (#9) followed by the Index Number.
- SETU ATA211G will check the **FXS Speed Dialing Table** to fetch the number configured for the Index number and the corresponding destination Destination trunk (SIP1 / SIP2 / SIP3 / Mobile) to dial out the number.

If you want to use Speed Dialing from the SIP Trunk,

- If you dial the SIP Trunk number of ATA211G.
- ATA will answer the call and then you can dial the feature access code for Speed Dial (#9) followed by the Index Number.
- SETU ATA211G will check the **SIP Speed Dialing Table** to fetch the number configured for the Index number.
- This number will be out dialed from the Mobile port.



*This feature will work on the SIP Trunk, only if Incoming Call Routing option is selected as **Answer the call, Collect number and route to Mobile**. See “[SIP Trunks](#)” topic.*

In the Number field, if you enter:

- *SIP ID, the Destination trunk configured can be a peer-to-peer or proxy SIP trunk.*
- *IP Address / SIP URI, the Destination trunk must be can a peer-to-peer SIP trunk only.*

If you want to use Speed Dialing from the Mobile port,

- Dial the Mobile port number of ATA211G.
- ATA will answer the call and then you can dial the feature access code for Speed Dial (#9) followed by the Index Number.
- SETU ATA211G will check the **Mobile Speed Dialing Table** to fetch the number configured for the Index number and the corresponding Destination trunk (SIP1 / SIP2 / SIP3) to dial out the number.



This feature will work on the Mobile port, only if Incoming Call Routing option is selected is:

- *Answer the call, collect number & route as per incoming call route selection.*
- *Route to FXS. If not answered within ring timer, answer the call, collect number & route as per incoming call route selection.*

Configuring Speed Dialing

To use Speed Dialing you must configure the Speed Dialing Table separately for each port: FXS, Mobile and SIP.

Speed Dialing on the FXS Port

From Admin mode:

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click **Speed Dialing** link to open the page. The Speed Dialing Table appears on this page.

You can program up to 20 entries in this Table from Index 01 to 20.

The screenshot shows the 'MATRIX SETU ATA211G' interface. On the left is a navigation menu with 'Basic Settings', 'Advanced Settings', 'Supplementary Services' (expanded), 'Supplementary Features', 'Speed Dialing' (selected), and 'Status'. The main area is titled 'Speed Dialing' and has tabs for 'FXS', 'Mobile', and 'SIP'. Below the tabs is a table with the following structure:

Index	Number	Name	Destination Route
01			SIP1
02			SIP1
03			SIP1
04			SIP1
05			SIP1
06			SIP1
07			SIP1
08			SIP1
09			SIP1
10			SIP1
11			SIP1
12			SIP1
13			SIP1
14			SIP1
15			SIP1
16			SIP1
17			SIP1

At the bottom of the table are two buttons: 'Submit' and 'Default'.

Against each index number:

- Enter the **Name** (Max 24 characters) of the contact you wish to speed dial.
 - Enter the **Number** (Max. 40 characters) against each name.
 - Select the appropriate **Destination Route**, through which you want the system to dial out the number.
4. Click **Submit** to save your entries.

From User mode:

1. Log in as **User**.
2. Click the **Speed Dialing** link. The Speed Dialing Table opens.

The screenshot shows the MATRIX SETU ATA211G web interface. The top header displays the MATRIX logo and the device name SETU ATA211G. On the left, a 'User Settings' sidebar lists options: Supplementary Features, Speed Dialing (selected), Network, User Password, Syslog, Factory Defaults, and PCAP. Below this is a 'Status' section. The main content area has tabs for FXS, Mobile, and SIP, with FXS selected. The 'Speed Dialing' section contains a table with 17 rows (Index 01 to 17). The table has four columns: Index, Number, Name, and Destination Route. The 'Destination Route' column shows 'SIP1' for all entries, with a dropdown arrow next to each. At the bottom of the table are 'Submit' and 'Default' buttons.

Index	Number	Name	Destination Route
01			SIP1
02			SIP1
03			SIP1
04			SIP1
05			SIP1
06			SIP1
07			SIP1
08			SIP1
09			SIP1
10			SIP1
11			SIP1
12			SIP1
13			SIP1
14			SIP1
15			SIP1
16			SIP1
17			SIP1

3. To configure the Speed Dialing Table, follow the same instructions as provided for configuring the table from the Admin mode.

Speed Dialing on the Mobile Port

From Admin mode:

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click **Speed Dialing** link to open the page. The Speed Dialing Table appears on this page.

You can program up to 20 entries in this Table from Index 01 to 20.


SETU ATA211G

Basic Settings

Advanced Settings

Supplementary Services

Supplementary Features

Speed Dialing

Status

FXS

Mobile

SIP

Speed Dialing

Index	Number	Name	Destination Route
01			SIP1
02			SIP1
03			SIP1
04			SIP1
05			SIP1
06			SIP1
07			SIP1
08			SIP1
09			SIP1
10			SIP1
11			SIP1
12			SIP1
13			SIP1
14			SIP1
15			SIP1
16			SIP1
17			SIP1

Submit

Default

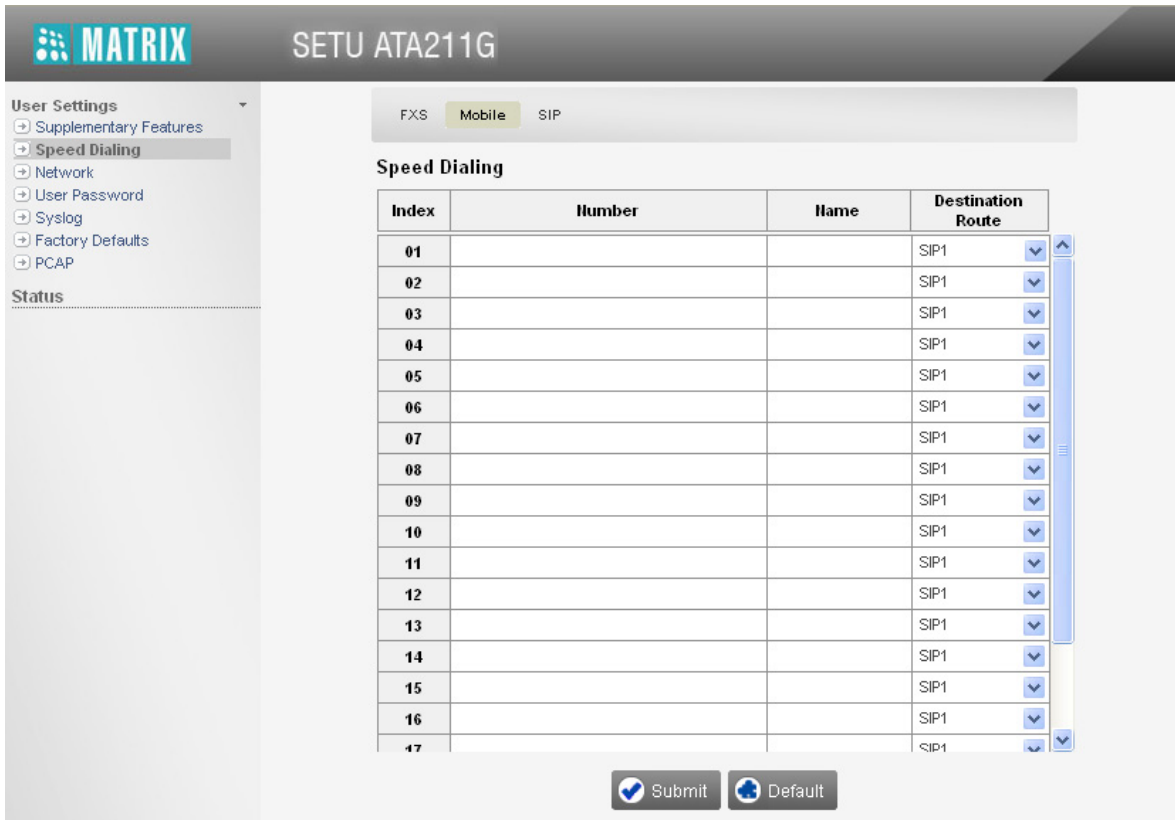
Against each index number:

- Enter the **Name** (Max 24 characters) of the contact you wish to speed dial.
- Enter the **Number** (Max. 40 characters) against each name.
- Select the appropriate **Destination Route**, through which you want the system to dial out the number.
You may select: SIP1 or SIP2 or SIP3. Default: SIP1.

4. Click **Submit** to save your entries.

From User mode:

1. Log in as **User**.
2. Click the **Speed Dialing** link. The Speed Dialing Table opens.



3. To configure the Speed Dialing Table, follow the same instructions as provided for configuring this table from the Admin mode.

Speed Dialing on the SIP Trunk

From Admin mode:

1. Log in as **Admin**.
2. Click the **Supplementary Services** link on the left pane to expand.
3. Click **Speed Dialing** link to open the page. The Speed Dialing Table appears on this page.

You can program up to 20 entries in this Table from Index 01 to 20.


SETU ATA211G

Basic Settings

Advanced Settings

Supplementary Services

Supplementary Features

Speed Dialing

Status

FXS Mobile SIP

Speed Dialing

Index	Number	Name	Destination Route
01			Mobile
02			Mobile
03			Mobile
04			Mobile
05			Mobile
06			Mobile
07			Mobile
08			Mobile
09			Mobile
10			Mobile
11			Mobile
12			Mobile
13			Mobile
14			Mobile
15			Mobile
16			Mobile
17			Mobile

Submit

Default

Against each index number:

- Enter the **Name** (Max 24 characters) of the contact you wish to speed dial.
- Enter the **Number** (Max. 40 characters) against each name.
- For each number and name you enter, the system will dial out the number through the **Mobile** port.

4. Click **Submit** to save your entries.

From User mode:

1. Log in as **User**.
2. Click the **Speed Dialing** link. The Speed Dialing Table opens.


SETU ATA211G

User Settings

- Supplementary Features
- Speed Dialing**
- Network
- User Password
- Syslog
- Factory Defaults
- PCAP

FXS
Mobile
SIP

Speed Dialing

Index	Number	Name	Destination Route
01			Mobile
02			Mobile
03			Mobile
04			Mobile
05			Mobile
06			Mobile
07			Mobile
08			Mobile
09			Mobile
10			Mobile
11			Mobile
12			Mobile
13			Mobile
14			Mobile
15			Mobile
16			Mobile
17			Mobile

- To configure the Speed Dialing Table, follow the same instructions as provided for configuring this table from the Admin mode.

Supplementary Services of Service Provider

When SETU ATA211G interfaced with a service provider server - ITSP, an ATA, the Matrix ETERNITY IP-PBX, or any other PBX - that supports supplementary services that require dialing of Flash¹², like Call Hold, Call Transfer, Call Waiting, etc., you may choose to access the features of the service provider, or to access the features of SETU ATA211G primarily.

If you want to use the features of the service provider, you must select **Network** as the *Subscriber Type* for the FXS port of SETU ATA211G.

If you want to use the features of SETU ATA211G, you must select **Gateway** as the Subscriber Type for the FXS port of SETU ATA211G. However, you will be able to access the features of the service provider in the Gateway mode, by dialing Flash, followed by **#0**, the Access Code for Using Supplementary Services of Service Provider.

When the SETU ATA211G is set in the Gateway mode, when you dial flash and the access code #0 to access the supplementary services of the service provider, the active call will be put on hold and you will get a feature tone. SETU ATA211G will start the an internal timer, called the **Service Provider Access Code Wait Timer**. This timer has a duration of 10 seconds. Any activity performed on the FXS port within this timer will be sent to the service provider server. Any activity performed on the FXS port after the expiry of this timer, will be processed by SETU ATA211G.



*To use Supplementary Services of Service Provider, configure the parameter **Subscriber Type** on the FXS port as **Network** or as **Gateway**. See “[FXS Port](#)” topic.*

Using Supplementary Services of Service Provider

If you have set SETU ATA211G in the Network Mode

- Dial **Flash** during speech.
You will get feature tone played by your service provider.
- Dial feature access code provided by the service providers server.

If you have set SETU ATA211G in the Gateway Mode

- Dial **Flash** during speech. This will place the active call on hold and you will get a feature tone.
- Dial **#0**.
You will be in Network mode for 10 seconds.
- Dial the feature access code provided by the service providers server within 10 seconds.

12. To be able to use features of the service provider SETU ATA211G supports dialing of Flash during speech on SIP.



- At the India office, PBX3 is interfaced with SETU ATA211G, which is assigned SIP ID 23.
- At the China office, PBX4 is interfaced with SETU ATA211G, which is assigned a SIP ID 24.
- Now, from the office in Canada, where PBX1 is installed, the extension user 101 of PBX1 wants to call extension 102 of PBX2 in Australia. This is how it would work **in absence of Multi-stage Dialing**:
 - Extension user 101 of PBX1 dials the access code for a trunk line, followed by the number 22, which is the SIP ID assigned to SETU ATA211G installed in Australia, and then the extension number 102. The PBX1 routes this call to its FXO port which is connected to the FXS port of SETU ATA - 21.
 - SETU ATA - 21, then routes this call to IP-PBX, which is further routed by the IP-PBX to SETU ATA - 22.
 - SETU ATA - 22 places the incoming call on its FXS port as per incoming call logic. PBX2 is connected to the FXS port of SETU ATA -22.
 - PBX2 receives the incoming call. PBX2 has Auto-Attendant or Direct Inward Dialing enabled on the landing trunk to answer the call on its FXO port.
 - Answer Signal is detected by SETU ATA -21.
 - PBX2 plays dial message/dial tone.
 - The caller (Extension 101 of PBX1) then dials 102, the desired called party number. PBX2 places the call to its extension,102.

Thus, to make a call to an extension of another office, extension users need to dial the number in two stages, each time, to reach the extension of the other office. In the first stage, the calling extension user dials the SIP ID of SETU ATA. In the second stage, the calling extension user dials the desired extension number.

When Multi-Stage Dialing is used, extension users need to dial only the complete number string in one stage, and SETU ATA211G converts the dialing method automatically into two stages.

To take the above example further, **when Multi-Stage Dialing is used** this is how it would work,

- SETU ATA -21 (at calling end) accepts the complete number—trunk access code + SIP ID of ATA in Australia + extension number 102—dialed by the extension user.
- It dials only the number of the remote SIP device, which is SETU ATA - 22 (at called side).
- When the call matures, SETU ATA - 21 (at calling end) then sends the remaining numbers - 102 - in DTMF to the called device, SETU ATA -22.
- While SETU ATA - 21 is sending the DTMF (102, the actual destination number which is the extension of called PBX 2), the caller (extension 101 of PBX1) will hear the routing tone, if the **Play Routing Tone** flag is enabled in the **System Parameters** of ATA - 21.

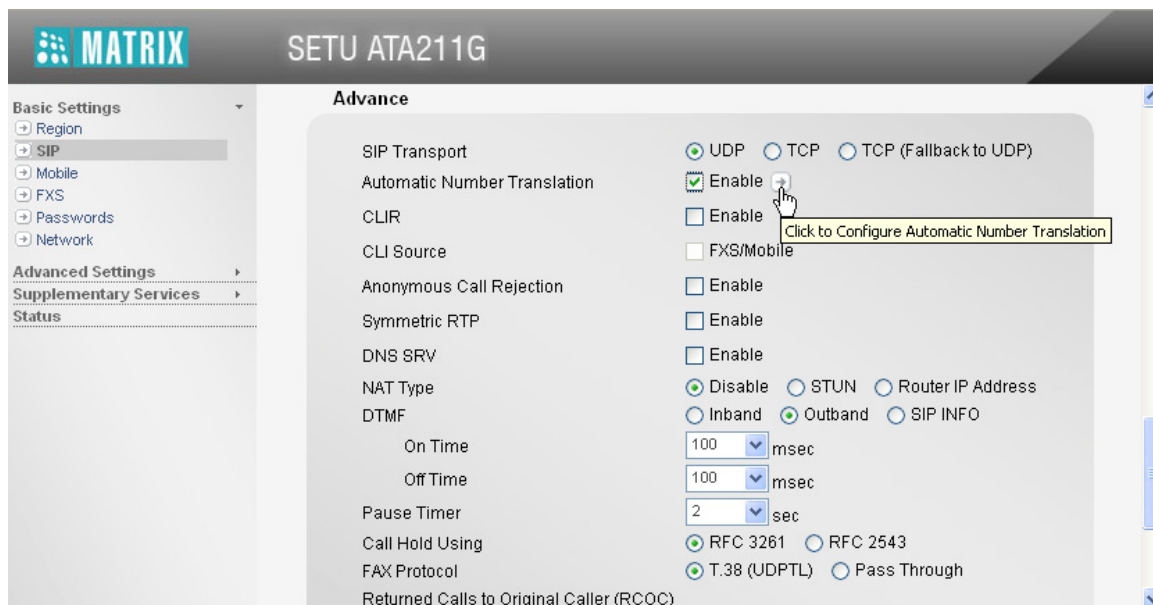
Configuring Multi-Stage Dialing

For this feature to work, at each site where the SETU ATA211G is installed, do the following:

- Make sure that Auto Attendant or Direct Inward Dialing is enabled on the FXO port of the PBX on which SETU ATA211G is interfaced.
- Enable Automatic Number Translation on the desired SIP Trunk of SETU ATA211G.
- Configure the Automatic Number Translation Table.
- Set the DTMF ON Time, DTMF OFF Time.
- Enable 'Play Routing Tone' in the System Parameters, if required.
- Set the Pause Timer to the desired value.

To configure Multistage Dialing at each site where the SETU ATA211G is connected,

- Open Jeeves.
- Log in as **Admin** with your password.
- Click the **Basic Settings** link.
- Under Basic Settings, click **SIP**.
- Now, click the SIP Trunk number tab (SIP Trunk 1, 2, or 3) on which you want to apply Multi-stage Dialing.
- Click the **More** button to view all parameters on this page.
- Scroll to **Advanced**.
- Enable **Automatic Number Translation** on this SIP Trunk.
- To configure the Automatic Number Translation Table, click the arrow icon.



The Automatic Number Translation Table opens in a new window.

Automatic Number Translation

Index	Dialed Number	Substitute Number
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		
20		
21		
22		
23		
24		

The Automatic Number Translation Table can accommodate upto 24 Dialed Numbers and their corresponding Substitute Numbers, which are stored against Index numbers 01 to 24.

In the **Dialed Number** column, enter the numbers that users will dial out using the SIP trunk.

In the **Substitute Number** column, enter the number which the system should dial out in place of the number dialed by the users.

The SETU ATA211G considers the character '~' as *Wait for Answer* and it considers the character '^' as *Pause*.

You may use these characters in the Substitute Numbers, if you want SETU ATA211G to provide suitable delay.

Make sure you enter a Dialed Number and its corresponding Substitute Number at the same Index. For example, if you enter 22 as Dialed Number at Index 01, enter the Substitute Number 22~ also at Index 01.

To take the above example further, if extensions of PBX1 want to reach the extensions of PBX2 by dialing 22 + Extension Number (of PBX2), you must configure the Dialed Number and Substitute Number at calling side, i.e. SETU ATA - 21, as follows:

Index	Dialed Number	Substitute Number
01	22	22~
02		
03		
::		
24		

When any extension user of PBX1 dials '22 + Extension Number (of PBX2)', for example, 22101 (to reach the extension 101 of PBX2),

- SETU ATA -21 will first dial out '22', which is the SIP ID of the called ATA and wait for answer' (22~). The caller will only hear routing tone.
- PBX2 on the called side answers the call (DID/Auto-Attendant must be enabled on the FXO port).
- When the call matures, SETU ATA - 21 will send the remaining numbers, 101 in the RFC 2833 (Outband) to the called device, SETU ATA - 22.
- These DTMF digits are then detected by PBX2 on its FXO port. The PBX places the call to extension 101.
- In the above scenario, if you want some delay after the call matures and before SETU ATA -21 sends the remaining DTMF digits, program '22~^' as Substitute Number instead of '22~'.

Index	Dialed Number	Substitute Number
01	22	22~^
02		
03		
::		
24		

- Click **Submit** to save your entries.
- Close the window.
- Keep the **DTMF ON** and **DTMF OFF Time** to default values, if you don't have any specific requirement.
- Change the **Pause Timer** to the desired value, if required. Default: 2 seconds.
- Click **Submit** to save your SIP Trunk settings.
- If you want Routing Tone to be played to the calling extension user, keep the Play Routing Tone flag enabled in the ["System Parameters"](#) page.
- You may log out of Jeeves or continue, as required.



The above example and configuration instructions are for Multi-stage Dialing on SIP trunks only. You may also configure Multi-stage Dialing on the Mobile port, if required.

End of Dialing

Unlike a normal phone which transmits each digit you press to the telephone exchange for dialing, when you are making an VoIP call, you must enter all the digits/characters and indicate to SETU ATA211G the 'End of Dialing', i.e. that you have finished dialing.

SETU ATA211G considers End of Dialing when:

1. The **Termination Digit, #** (non-programmable), is pressed after dialing the number/IP Address/SIP URI.
2. The **Inter Digit Wait Timer** elapses, after which the system automatically dials out the number. By default the Inter Digit Timer in the system is set to 5 seconds. The system will wait for 5 seconds after the last digit is pressed, consider the number as complete and dial it out automatically.

To change the duration of the Inter Digit Wait Timer, see [“System Parameters”](#).

3. The number of digits dialed from the phone connected to the FXS port matches with the length of the **Maximum Digits** configured in the [“Dialed Number Table”](#). Default: 16 digits.

When you dial a number from the phone connected to SETU ATA, the system will match it with the above end-of-dialing indications and accept the one that matches first.

If you do not want SETU ATA to wait for the Inter Digit Wait Timer to expire to dial out the number, you may dial the termination digit '#' after dialing the number string.



The system will consider the Maximum Digits configured as End-of-Dialing only

- when **Route all calls as per Dialed Number Table** is selected as **Outgoing Call** routing option for the FXS port.
- when **Dialed Number Table** is selected as **Incoming Call Route** for the Mobile port.

Mobile Network Selection

When SETU ATA211G is located in the home region, then, the SIM card gets registered with the Network Operator of that SIM only, leading to no extra charges to the user (i.e. no roaming charges). Whereas, if SETU ATA211G is located in an area away from the home region or in an area where there are more than one Network Operators are available, then, the SIM card may get registered with another available Network and result in **roaming** charges.

Thus, **Mobile Network Selection** feature of SETU ATA211G enables you to select the GSM network of your own choice, either **automatically** or **manually**. At Power ON, ATA211G will automatically locate and register the SIM with the Network that supports the SIM card installed in it. If the SIM fails to register, it will restart the process of network selection on the expiry of the **Network Registration Retry Timer**.

Types of Network Selection Mode

SETU ATA211G gives you the flexibility to select network selection mode as per your requirement. ATA211G supports two modes of network selection: **Automatic** and **Manual**. Before you enable the Manual mode for Network Selection, Network Operator Table is configured, consisting of **Network Operator Codes** and its priority.

Automatic

This mode is selected when you require that your mobile station should always get registered with the available GSM network whether you are in home region or outside the home region, to make and receive calls at any cost.

- In Automatic mode, mobile port gets registered with any available network whose signal strength is more compared to other available networks.

Manual

This mode is selected when you always wish to get registered with the local network operator available in the region.

- If ATA211G is installed in the border area where more than one GSM network operator is available and network selection mode is set to automatic then it may happen that the SIM is registered with the other region's network operator. This is undesirable. Hence, in this situation, **Manual** mode of network selection is selected.

If you select the Mobile Network Selection as **Manual**, you must configure Network Operator Codes (MCC and MNC) according to priority you want to assign to each, from **Priority 1** to **Priority 9**.

With Network Operator Codes (**MCC and MNC**) configured according to their priority, at every Power on, ATA211G will send a request for available GSM Network Operators, who in turn will respond to the request with a list of available Network Operator Codes. ATA211G will match the available Network Operator Codes with those you configured and select the Network Operator which best suits your requirement in order of priority.

- If the Mobile Port fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer.
- If no match is found, the SIM will not get registered with any of the available Network Operators and no calls can be made or received on this port.

About Network Operator Codes

Network Operator Codes comprises of **MCC (Mobile Country Code)** appended by **MNC (Mobile Network Code)**.

- **MCC:** It identifies a country and a country may be assigned more than one MCC. For example, MCC assigned to India is 404 which is same for all the network operators in India. MCCs are generally of 3 digits.
- **MNC:** It is a Mobile Network Code and is generally of 2-3 digits.

The MCC-MNC combination uniquely identifies the home network of the mobile terminal or mobile user. MNC are different in each state for same network operator. For example, Airtel, a GSM network operator in India has different MNC assigned to its networks in various states. The MNC for Airtel in Maharashtra state is 90, whereas, the same for the Gujarat state is 98.

To set Network Selection Mode, see [“Mobile Port”](#).

Return Call to Original Caller (RCOC)

When an outgoing call made from SETU ATA211G is returned by the called party, the call lands on the port from which the call was made. The call is then routed according to the incoming call route selection configured in SETU ATA211G. Now, it is desirable that the returned call be routed to the port that originally made the call. However, if the port that originally made the call is not configured as the destination for incoming call routing, the returned call will be routed to a different port instead of the port that originally made the call.

With the Return Call to Original Caller (RCOC) feature, regardless of the incoming call route selection configured in the system, it is possible to route the call returned by the called party directly to the same port from which the call was originally made.

How it works?

For this feature to work, it must be enabled on the desired port(s), i.e. the Mobile Port, SIP Trunks.

SETU ATA211G supports three types of RCOC:

1. **RCOC - On Busy:** When an outgoing call is made from SETU ATA211G, RCOC is set when the called party is found to be busy.
2. **RCOC - On Speech:** When an outgoing call is made from SETU ATA211G, RCOC is set when the called party answers the call.
3. **RCOC - On No Reply:** When an outgoing call is made from SETU ATA211G, RCOC is set when the called party does not answer the call.

You can enable any or all three types of RCOC on the Mobile port and SIP trunks.

Here is an example of how this feature works:

- An outgoing call is made from the FXS port (Source Port) of SETU ATA211G.
- The system checks the Outgoing Call Route, and finds the Mobile port as the Destination Port.
- The system checks for the type of RCOC enabled on the Mobile port (Destination Port).
- If RCOC-On Busy is enabled,
 - If the called party is found busy, the system stores the record of the outgoing call in an internal database referred to as the RCOC Table.
- If RCOC-On Speech is enabled,
 - when the called party answers the call, the system stores the record of the outgoing call in the RCOC Table.
- If RCOC-No Reply is enabled,
 - when there is no response from called party, the system stores the record of the outgoing call in the RCOC Table.
- The system stores the following details of the outgoing call in the RCOC Table:

- *Calling Party Number:* The number received on the Source Port. In this example, it is the number configured for the FXS port.
- *Called Party Number:* The number received in the destination port (Mobile Port/SIP Trunk) from the Source Port. In this example, it is the number dialed from the FXS port and received on the Mobile Port.
- *Source Port Type and Port Number:* The number of the Source port from which the outgoing call was originally made. In this example, the FXS port.

The RCOC Table can store up to 200 such entries.

- The system stores the details of each outgoing call for the duration of the RCOC Delete Record Timer, which is configurable. By default, the Timer is set to 999 minutes. On the expiry of this Timer, the details are deleted.
- Now, when there is an incoming call on the Mobile port/SIP trunk on which RCOC is enabled, the system matches the CLI of the incoming call with the RCOC Table.
- For example, if incoming call is received on the Mobile port, then, CLI received on Mobile Port is checked with the Called Party Number stored in the RCOC table.

The system uses the best match found logic. It matches 7 digits of the CLI number with the digits of the Called Party Number, from the right hand side. For example: +919824284984 is stored in the RCOC table, the incoming CLI is 9824284984, the system will match 7 digits from the right hand side 4284984 to arrive at the best match.

- When a match is found, the system checks the corresponding Calling Party Number and Source port stored for the Called Party Number stored in the RCOC Table, and routes the call to the Source port, i.e. from which the call was originally made.
- The system clears the record of this call from the RCOC Table.
- If no match is found in the RCOC Table, the call will be routed according to the Incoming Call Route selection configured in the system.



- *If the RCOC Record Delete Timer has expired when the call is returned, the call will be routed as per the Incoming Call Route selection, because no matching record exists in the RCOC Table.*
- *Records of outgoing calls made using the features Call Transfer-Blind, Call Transfer-Attended, and Call Forward, will not be stored in the RCOC table.*
- *The RCOC Table can store records of 200 outgoing calls. You can delete all the entries in the table, any time you want.*

Configuring RCOC feature

1. Log in as an **Admin**.
2. Click **Basic Settings** to expand.

3. Click **Mobile** link or **SIP** link as per your requirement to enable RCOC feature on these ports.

The screenshot displays the 'Advance' configuration page for the Matrix SETU ATA211G. The left sidebar contains navigation links: 'Basic Settings' (with sub-links for Region, SIP, Mobile, FXS, Passwords, and Network), 'Advanced Settings', 'Supplementary Services', and 'Status'. The main content area is titled 'Advance' and contains several configuration sections. The 'Returned Calls to Original Caller (RCOC)' section is highlighted with a red rectangular box. This section includes three sub-headers: 'On Busy', 'On No Response', and 'On Speech', each followed by an 'Enable' checkbox. Other visible settings include 'SIP Transport' (UDP selected), 'Automatic Number Translation' (Enable button), 'CLIR' (checked), 'CLI Source' (FXS/Mobile), 'Anonymous Call Rejection' (Enable), 'Symmetric RTP' (Enable), 'DNS SRV' (Enable), 'NAT Type' (Disable selected), 'DTMF' (Inband selected), 'On Time' (100 msec), 'Off Time' (100 msec), 'Pause Timer' (2 sec), 'Call Hold Using' (RFC 3261 selected), 'FAX Protocol' (T.38 (UDPTL) selected), and various other options like STUN, Outband, Router IP Address, SIP INFO, RFC 2543, and Pass Through.

4. Now, select the type of RCOC you want to enable on the Mobile Port/SIP Trunk.

On Busy:

- Select the **Enable** check box if you want the returned call to be routed to the original port if the called party is found to be busy.

On No Response:

- Select the **Enable** check box if you want the returned call to be routed to the original port if the called party does not answer the call.

On Speech:

- Select the **Enable** check box if you want the returned call to be routed to the original port if the called party answers the call.

5. Click **Submit** to save.

6. Now, to configure RCOC Record Delete Timer, click **System** link under **Advance Settings** link.

MATRIX SETU ATA211G

Basic Settings ▶

Advanced Settings ▼

- System
- Called Party Number Table
- Dialed Number Table
- Peer-to-Peer Numbers
- Mobile Emergency Numbers
- Digest Authentication
- PIN Authentication
- Daylight Saving Time
- Static Routing
- Factory Defaults
- Restart
- Syslog
- PCAP
- Software Upgrade
- Auto-Configuration

Supplementary Services ▶

Status

General

System Name: SETU ATA211G

First Digit Wait Timer: 15 sec

Inter Digit Wait Timer: 5 sec

Transfer Notification Timer: 60 sec

Play Routing Tone: ☒ Enable

Call Release Timer: ☒ Enable

Timer: 999 min

VoIP Silence Disconnect Timer: ☒ Enable

Timer: 99 min

RCOC Record Delete Timer: 999 min

Clear RCOC Table

Forward Error Correction: ☐ Enable

Voice Activity Detection: ☐ Enable

Default Admin Password when User Defaults the ATA: ☒ Yes

NAT

STUN Server Address:Port: : 03478

SIP Port Fetched Using STUN: ☒ Enable

RCOC Record Delete Timer:

- In **RCOC Record Delete Timer** field, enter the time (in minutes) after which the entry stored in RCOC Table should be deleted automatically.

RCOC Record Delete Timer starts as soon as the called party is found busy, or is not responding, or answers the call (depending on the type of RCOC set) and the system stores the record of the outgoing call in the RCOC Table.

After the expiry of RCOC Record Delete Timer, the entry stored in the database is deleted automatically by the system. The range of this timer is 001 - 999 minutes. By default, it is set to **999** minutes.

- Clear RCOC Table:** Click this button, whenever you want all the entries of the RCOC table to be cleared.

7. Click **Submit** button to save.

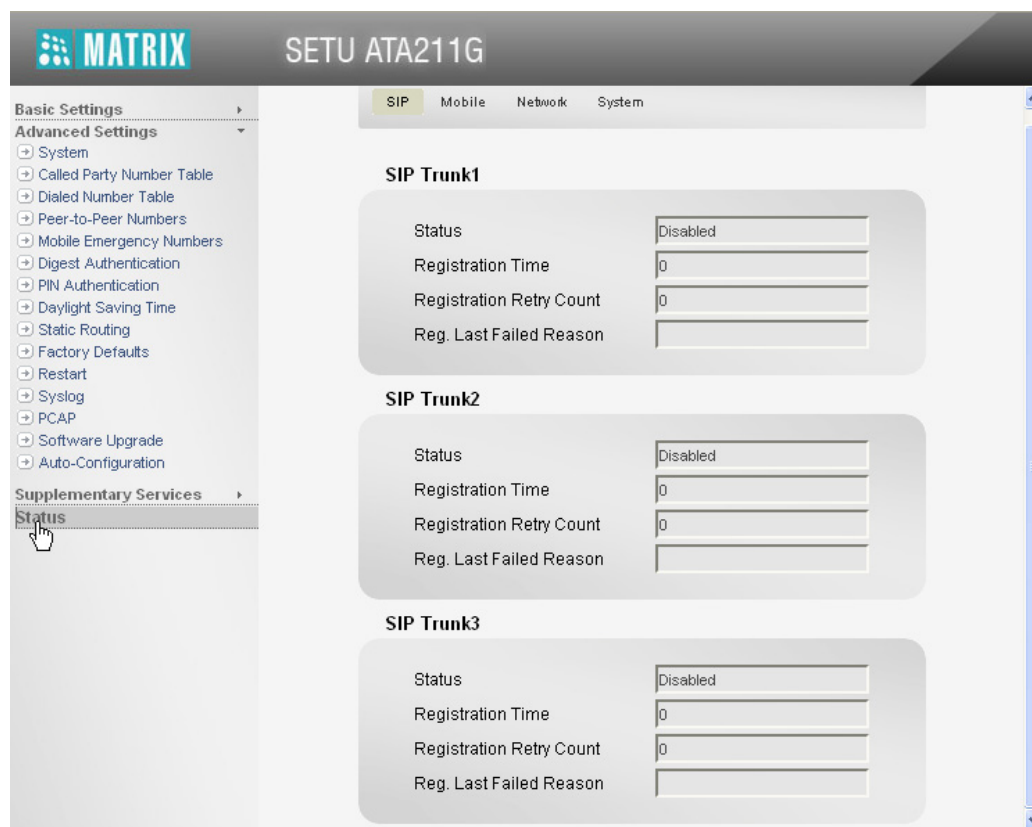
You may view the status of the System, the Network ports (LAN and WAN), SIP Trunks and Auto Configuration from Jeeves.

You can also check the status of certain parameters, like the Registration status of SIP Trunks, IP Address and Subnet Mask of the LAN and WAN port, on the display of the Phone connected to FXS port by dialing the relevant [“System Commands”](#).

Viewing System Status from Jeeves

To view System Status on Jeeves from Admin Mode,

- Log in as **Admin**.
- Click the **Status** link to open the page.



To view System Status on Jeeves from User Mode,

- Log in as **User**.
- Click **Status** link to open the page.

The screenshot shows the MATRIX SETU ATA211G web interface. On the left is a sidebar with a 'User Settings' dropdown menu containing links to Supplementary Features, Speed Dialing, Network, User Password, Syslog, Factory Defaults, and PCAP. Below this is a 'Status' link with a mouse cursor pointing to it. The main content area has a top navigation bar with 'SIP', 'Mobile', 'Network', and 'System' tabs. Below this, there are three sections for SIP Trunk1, SIP Trunk2, and SIP Trunk3. Each section contains four fields: Status (set to 'Disabled'), Registration Time (set to '0'), Registration Retry Count (set to '0'), and Reg. Last Failed Reason (empty).

SIP

The following parameters will be displayed for SIP Trunks 1, 2 and 3.

- **Status:** In this field, different status messages may be displayed, each of which are described briefly in the table below.

Status	Description
Disable	Shows that SIP Trunk is disabled.
Registering	Shows that SIP Trunk is enabled and is waiting for response from the SIP server.
Active	Shows that SIP Trunk is registered with the SIP server.
Failed	Shows that some error has occurred in the SIP Trunk and no calls can be made using it (applicable only in case of Proxy Account).

- **Registration Time:** The SIP Trunk is registered with the Registrar Server for a particular time period, after which it has to be re-registered. The registrar server indicates the time remaining for re-registration of the SIP Trunk. The same time is displayed in this field.

- **Registration Retry Count:** This field displays the total number of register messages which have been sent to the registrar server for registering SIP Trunk.
- **Reg. Last Failed Reason:** This field displays the reason for failure of SIP Trunk registration with the registrar server. The different reasons for registration failure that may appear in this field are:

Reason for Failure	Description
Message send fail	This reason is displayed when registration request sent to registrar server fails.
Failed to create Register client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ the number of SIP clients to register is greater than the number programmed in the stack.
Failed to detach register client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ the number of SIP clients to register is greater than the number programmed in the stack.
Failed to send request	This reason is displayed when DNS server is not programmed.
Local Failure	This reason is displayed when DNS query fails.
Response time-out	This reason is displayed on the expiry of the General Request Timer.
Error Response- 4xx to 6xx	This is error response code.
No contact header in 2xx	This reason is displayed when no contact address is received in 2xx response from the SIP server.
Authentication Failed	This reason is displayed when the SIP server does not authenticate SETU ATA.
STUN address is not programmed	This reason is displayed when STUN is enabled but address is not configured.
STUN query fail	This reason is displayed when a query to the STUN server fails.
Outbound address is not programmed	This reason is displayed when Outbound is enabled but Outbound address is not configured.
Router's IP address is not programmed	This reason is displayed when Router's IP Address is to be used in signaling but the address is not programmed.

Mobile

MATRIX SETU ATA211G

SIP **Mobile** Network System

Mobile

IMEI Number	354779032492770
Currently Registered With (Network Operator Code)	Not Registered
Currently Registered With (Network Operator Name)	
Signal Strength	095 -dBm
Activity Status	SIM Absent
Firmware Version of Engine	1604B02SIM340M32_SST34HF

Following parameters will be displayed for the Mobile port.

- **IMEI Number:** This field displays the IMEI number¹³ of the GSM network.
- **Currently Registered With (Network Operator Code):** This field displays the code of the currently registered network with which the Mobile Port is registered. If it is not registered with any of the network types then the system displays 'not registered' status. Default: Not Registered.
- **Currently Registered With (Network Operator Name):** This field displays the name of the network with which the Mobile Port is currently registered with.
- **Signal Strength:** This field displays the current signal strength.
- **Activity Status:** This field displays the below status of the port's activity:
 - Module Initialization
 - SIM PUK Required
 - SIM PIN Wrong
 - SIM Absent
 - SIM Present
 - Network Absent
 - Network Present
- **Firmware Version of Engine:** This field displays the current firmware's version as well as revision of engine.

Network Parameters

Following status of network parameters (LAN and WAN) will be displayed on the Network page.

MATRIX SETU ATA211G

SIP Mobile **Network** System

LAN Port

IP Address	192.168.2.21
Subnet Mask	255.255.255.0
MAC Address	00:50:c2:55:b0:20

WAN Port

Routing Mode	NAT
IP Address	192.168.1.196
Subnet Mask	255.255.255.0
Gateway Address	
Primary DNS Address	
Secondary DNS Address	
DNS Domain Name	
MAC Address	00:50:c2:55:b0:28
NAT Type	UNKNOWN
IP Address Fetched Using STUN	
SIP Port Fetched Using STUN	

13. IMEI (International Mobile Equipment Identity) Number is a unique identity number of the GSM engine.

LAN

- **IP Address:** This field displays the current IP Address of the LAN port.
- **Subnet Mask:** This field displays the current Subnet Mask Address of the LAN port.
- **MAC Address:** This field displays the MAC Address of the LAN port.

WAN

- **Routing Mode:** This field displays the Routing Mode selected, that is NAT or Bridge.
- **IP Address:** This field displays the current IP Address of the WAN port.
- **Subnet Mask:** This field displays the current Subnet Mask Address of the WAN port.
- **Gateway Address:** This field displays Gateway Address assigned to the WAN Port of SETU ATA211G. This field displays the IP Address of the LAN port of the Router.
- **Primary DNS Address:** This field displays the Primary DNS address of SETU ATA211G.
- **Secondary DNS Address:** This field displays the Primary DNS address of SETU ATA211G.
- **DNS Domain Name:** This field displays the DNS Domain Name of SETU ATA211G.
- **MAC Address:** This field displays MAC Address of the WAN Port of SETU ATA211G.
- **NAT Type:** This field displays the NAT Type, if STUN is enabled in SETU ATA211G. Commonly used NAT types are:
 - Unknown
 - Open
 - Conenat
 - Restrictednat
 - Portrestrictednat
 - Symmetricnat
 - Symmetricfirewall
 - Blocked
- **IP Address fetched using STUN:** This field displays the IP address fetched using STUN, if STUN server address is configured.
- **SIP Port fetched using STUN:** This field displays the SIP port fetched using STUN if STUN server address is configured.

System

Following status of System parameters are displayed on this page.

MATRIX SETU ATA211G

SIP Mobile Network **System**

Date & Time

Current Date: 02 February 2011
Current Time: 17:32:49

Firmware

Version/Revision: V2R1
Kernel Date: 03-Apr-2010

Auto Configuration

Status: Never
Next Resync:
Last Resync:

Date & Time

- **Current Date:** Current Date is displayed in this field.
- **Current Time:** Current Time is displayed in this field.

Firmware

- **Version-Revision:** Current Software Version-Revision of SETU ATA211G is displayed here.
- **Kernel Date:** The Date of current Kernel of the system is displayed here.

Auto Configuration

- **Status:** This field displays the type of Auto Configuration option selected.
- **Next Resync:** This field displays when the system will resync next for Auto Configuration.
- **Last Resync:** This field displays the time when the system last resynchronized with the Auto Configuration server.

You may log out of Jeeves.

Viewing System Status on Phone

To view System Status on the phone's display from Admin Mode,

- Lift the receiver of the phone.
- Dial **#19**.
- Dial the Admin Password (default: **1234**).

To view System Status on phone's display from User Mode,

- Lift the receiver of the phone.
- Dial **#18**.
- Dial the User Password (default: **1234**).

You will get programming tone.

On entering Admin/User Mode,

- Dial the System Command for the status you want to view.
- You get confirmation tone.
- Go On-hook (replace the receiver of the phone) on confirmation tone.
- Your phone will ring.
- The status for the port or trunk for you dialed the command will be displayed on your phone.

For example:

To view the IP address of the WAN port,

Dial **31-#***

Go On-hook at the confirmation tone.

Your phone rings, and the IP Address will be displayed on your phone as: 192168001196

To view the WAN Subnet Mask,

Dial **33-#*** and go On-hook at the confirmation tone.

Your phone rings and WAN Subnet Mask is displayed as: 255255255000

To view the IP address of LAN port,

Dial **32-#***

Go On-hook at the confirmation tone.

Your phone rings, and the IP Address of the LAN port is displayed as: 192168002021

To view the LAN Subnet Mask,

Dial **34-#*** and go On-hook at the confirmation tone.

Your phone rings and the LAN Subnet mask is displayed as: 255255255000

To view the Gateway Address,

Dial **35-#*** and go On-hook at the confirmation tone.
Your phone rings and the Gateway Address is displayed as: 192168001254

To view DNS Address,
Dial **36-#*** and go On-hook at the confirmation tone.
Your phone rings and the DNS Address is displayed as: 202100098054

To view the Connection Type,
Dial **37-#*** and go On-hook at the confirmation tone.

If your connection type is DHCP, the phone will display 001
If your connection type is PPPoE, the phone will display 002
If your connection type is Static, the phone will display 003

To view the registration status of SIP Trunk,
Dial **38-SIP Trunk Number-#*** and go On-hook at confirmation tone.

SIP Trunk Number is 1 for SIP Trunk1, 2 for SIP Trunk2 and 3 for SIP Trunk3

Your phone will ring and display one of the following:
001 for Disabled
002 for Registered
003 for Registering, when registration request is sent and the system is waiting for response
004 if the SIP trunk is configured as Peer-to-Peer Trunk.
005 if the registration request has failed, cannot be sent to the server.



Each time you go On-hook to view status of a port or trunk, you must enter the configuration mode again to view the status of another port or trunks.

Appendix

Call Progress Tone Generation

Code	Country	Dial Tone		Ring Back Tone		Busy Tone		Error Tone1		Error Tone2	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
		Hz	second	Hz	second	Hz	second	Hz	second	Hz	second
01	Australia	425*25	cont.	400*25	0.4on 0.2off 0.4on 2.0off	425	0.375on 0.375off	425	0.375on 0.375off	425	1on 1off
02	Argentina	425	cont.	425	1.0on 4.0 off	425	0.3on 0.2off	425	0.3on 0.4off	425	1on 1off
03	Belgium	425	cont.	425	1.0on 3.0off	425	0.5on 0.5off	425	0.167on 0.167off	425	1on 1off
04	Brazil	425	cont.	425	1.0on 4.0 off	425	0.25on 0.25off	425	0.25on 0.25 off	425	1on 1off
05	China	450	cont.	450	1.0on 4.0off	450	0.35 on 0.36off	450	0.7on 0.7off	450	1on 1off
06	Egypt	425*50	cont.	425*50	2.0on 1.0off	425*50	1.0on 4.0off	450	0.5on 0.5off	450	1on 1off
07	France	440	cont.	440	1.5on 3.5off	440	0.5on 0.5off	440	0.25on 0.25off	440	1on 1off
08	Germany	425	cont.	425	1.0on 4.0off	425	0.48on 0.48off	425	0.24on 0.24off	425	1on 1off
09	Greece	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.3on 0.3off	425	0.15on 0.15off	425	1on 1off
10	India	400*25	cont.	400*25	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off	400	0.25on 0.25off	400	1on 1off
11	Indonesia	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	425	1on 1off
12	Iran	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	425	1on 1off
13	Israel	400	cont.	400	1.0on 3.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	1on 1off
14	Italy	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.2on 0.2off	425	1on 1off
15	Japan	400	cont.	400*20	1.0on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	1on 1off
16	Kenya	425	cont.	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off 0.2on 0.6off	425	0.2on 0.6off	425	1on 1off
17	Korea	350+440	cont.	440+480	1.0on 2.0off	480+620	0.5on 0.5off	480+620	0.3on 0.2off	480+620	1on 1off
18	Malaysia	425	cont.	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.5off	425	0.5on 0.25off	425	1on 1off
19	Mexico	425	cont.	425	1.0on 4.0off	425	0.25on 0.25off	425	0.25on 0.25off	425	1on 1off
20	New Zealand	400	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	1on 1off
21	Philippines	425	cont.	425+480	1.0on 4.0off	480+620	0.5on 0.5off	480+620	0.25on 0.25off	480+620	1on 1off
22	Poland	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.5on 0.5off	425	1on 1off
23	Portugal	425	cont.	425	1.0on 5.0off	425	0.5on 0.5off	450	0.33on 1.0off	450	1on 1off
24	Russia	425	cont.	425	0.8on 3.2off	425	0.4on 0.4off	425	0.25on 0.25off	425	1on 1off
25	Saudi Arabia	425	cont.	425	1.2on 4.6off	425	0.5on 0.5off	425	0.25on 0.25off	425	1on 1off
26	Singapore	425	cont.	425*24	0.4on 0.2off 0.4on 2.0off	425	0.75on 0.75off	425	0.25on 0.25off	425	1on 1off
27	South Africa	400*33	cont.	400*33	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	1on 1off
28	Spain	425	cont.	425	1.5on 3.0off	425	0.2on 0.2off	425	0.25on 0.25off	425	1on 1off
29	Thailand	400*50	cont.	400	1.0on 4.0off	400	0.5on 0.5off	400	0.3on 0.3off	400	1on 1off
30	Turkey	450	cont.	450	2.0on 4.0off	450	0.5on 0.5off	450	0.2on 0.2off 0.6on 0.2off	450	1on 1off
31	UAE	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off	400	0.4on 0.35off 0.225on 0.525off	400	1on 1off

Code	Country	Dial Tone		Ring Back Tone		Busy Tone		Error Tone1		Error Tone2	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
		Hz	second	Hz	second	Hz	second	Hz	second	Hz	second
32	UK	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off	400	0.4on 0.35off 0.225on 0.525off	400	1on 1off
33	USA/Canada	350+440	cont.	440+480	2.0on 4.0off	480+620	0.5on 0.5off	480+620	0.25on 0.25off	480+620	1on 1off

Note1:

Cont. = continuous

f1*f2: f1 is modulated by f2.

f1+f2: the juxtaposition of two freq. f1 and f2 without modulation.

Note 2:

The frequency and cadence for following tones are same and fixed for all countries:

- Confirmation Tone: 400Hz, 0.1 on - 0.1 off
- Programming Tone: 400Hz, 0.1 on - 0.9 off
- CCWT: 400Hz, 0.3 on 6.0 off
- Routing Tone: 400Hz, 0.1 on 1.9 off

Ring Cadences

Code	Country	Frequency (Hz)	CADENCE (In Seconds)			
			T1ON	T1OFF	T2ON	T2OFF
01	Australia	25	0.4	0.2	0.4	2.0
02	Belgium	25	1.0	3.0		
03	Brazil	25	1.0	4.0		
04	China	25	1.0	3.0		
05	Egypt	25	2.0	4.0		
06	France	25	1.5	3.5		
07	Germany	25	3.5	5.5	0.79	1.1
08	Greece	25	1.0	4.0		
09	India	25	0.4	0.2	0.4	2.0
10	Israel	25	2.0	3.0		
11	Italy	25	1.0	4.0		
12	Japan	25	1.0	2.0		
13	Korea	25	1.0	3.0		
14	Malaysia	25	2.0	3.0		
15	New Zealand	25	2.0	3.0		
16	Poland	25	2.0	3.0		
17	Portugal	25	1.0	5.0		
18	Russia	25	1.0	3.0		
19	Singapore	25	0.4	0.2	0.4	2.0
20	South Africa	25	0.4	0.2	0.4	2.0
21	Spain	25	1.5	3.0		
22	Thailand	25	2.0	3.0		
23	UAE	25	2.0	3.0		
24	UK	25	0.4	0.2	0.4	2.0
25	USA/Canada	25	2.0	4.0		

Code	Country	Frequency (Hz)	CADENCE (In Seconds)			
			T1ON	T1OFF	T2ON	T2OFF
01	Australia	25	0.4	0.2	0.4	2.0
02	Belgium	25	1.0	3.0		
03	Brazil	25	1.0	4.0		
04	China	25	1.0	3.0		
05	Egypt	25	2.0	4.0		
06	France	25	1.5	3.5		
07	Germany	25	3.5	5.5	0.79	1.1
08	Greece	25	1.0	4.0		
09	India	25	0.4	0.2	0.4	2.0
10	Israel	25	2.0	3.0		
11	Italy	25	1.0	4.0		
12	Japan	25	1.0	2.0		
13	Korea	25	1.0	3.0		
14	Malaysia	25	2.0	3.0		
15	New Zealand	25	2.0	3.0		
16	Poland	25	2.0	3.0		
17	Portugal	25	1.0	5.0		
18	Russia	25	1.0	3.0		
19	Singapore	25	0.4	0.2	0.4	2.0
20	South Africa	25	0.4	0.2	0.4	2.0
21	Spain	25	1.5	3.0		
22	Thailand	25	2.0	3.0		
23	UAE	25	2.0	3.0		
24	UK	25	0.4	0.2	0.4	2.0
25	USA/Canada	25	2.0	4.0		

Where,

T1ON, T1OFF, are ON and OFF period for first ring cadence and T2ON, T2OFF, are ON and OFF period for second ring cadence.

Waveform for 'ring', is Sine Wave.

Frequently Asked Questions (FAQs)

Q.1 Which internet telephony (VoIP) protocols are supported by ATA211G?

Ans. ATA211G supports Session Initiation Protocol (SIP) to make/receive calls on the Internet.

Q.2 Can I make a call to another SIP terminal (e.g. Soft Phone running on PC) in my LAN network?

Ans. Yes. You can make a call to any SIP terminal in your LAN just by dialing an IP address of the SIP terminal. If you are not comfortable in dialing an IP address, you can use 'Peer to Peer' feature which allows you to dial the number instead an IP address for such type of applications.

Q.3 I have only one Internet connection for Internet surfing. Do I need another internet connection for making VoIP calls using ATA211G?

Ans. No. You do not need another internet connection to use ATA211G. ATA211G has inbuilt router which enables you to surf internet and making VoIP calls at the same time.

Q.4 I am using the ATA211G to make VoIP calls and surf internet. I have connected my PC to LAN port

of ATA211G. Can I still surf the internet when ATA211G is power down?

Ans. No. You can not surf Internet if ATA211G is power down. You must power ON the ATA211G to enable Internet surfing on your PC. Else, connect the PC directly to the internet.

Q.5 I am using the ATA211G to make VoIP calls and surf internet. I have connected my PC to LAN port of ATA211G. Can I still make VoIP calls when my PC is shut down or not connected?

Ans. Yes. You can still make VoIP calls when your PC is shutdown or not connected.

Q.6 My Internet Service Provider (ISP) supports only PPPoE for internet services. Can I use ATA211G with my ISP?

Ans. Yes. You need to configure PPPoE User-Id and PPPoE Password in ATA211G provided by your ISP.

Q.7 My internet Service Provider (ISP) supports only DHCP to provide internet services. Can I use ATA211G with my ISP?

Ans. Yes. You need to enable the DHCP client in ATA211G.

Q.8 My internet service provider (ISP) provides me only dynamic IP address. Can I use ATA211G to make VoIP calls and surf Internet at the same time?

Ans. Yes. Enable DHCP or PPPoE client in ATA211G as used by your ISP to provide internet services.

Q.9 I do not have PC to access Web Jeeves for configuration. Can I change the IP address and subnet mask of LAN/ WAN port?

Ans. Yes. You can change the IP address and subnet mask of LAN/ WAN port of ATA211G if you do not have PC to access Web Jeeves. You need to issue commands from the FXS port to change the IP address and subnet mask.

Q.10 I do not have PC to access Web Jeeves for configuration. Can I change the internet connection type of ATA211G to DHCP or PPPoE or Static?

Ans. Yes. You can change the connection type if you do not have PC to access Web Jeeves. You need to issue commands from the FXS port to change the connection type.

Q.11 I do not have PC to access Web Jeeves for configuration. Can I know whether ATA211G is registered with ITSP or not?

Ans. Yes. You can know the status of your SIP registration using LED provided on the enclosure of ATA211G or by command to display registration status on the Phone connected to FXS port.

Q.12 I have connected my PC to LAN port of ATA211G. But, I am not able to access the Web Jeeves for configuration.

Ans. Ensure that the IP address of LAN port of ATA211G and IP address of your computer should be in same subnet to enable Web Jeeves from your computer. If both the IP address is in the same subnet, check for any error in the Ethernet cable which connects your computer to LAN port of ATA211G.

Q.13 Can I upgrade my ATA211G for new firmware?

Ans. Yes. You can upgrade the ATA211G for new firmware. To upgrade ATA211G for new firmware, refer [“Software Upgradation”](#).

Q.14 Can I upgrade the ATA211G for new firmware remotely?

Ans. Yes. If you know the public IP address of WAN port of ATA211G, you can use the FTP server of ATA211G to upgrade the firmware. Refer [“Software Upgradation”](#).

Q.15 If I forgot the password, can I still use the Web Jeeves for configuration?

Ans. Yes. But you need to set the password value to factory default.

There are two important passwords for the configuration of ATA211G viz Admin Password and User Password. Admin password protects the access to whole configuration while User password protects the access to only few configuration of ATA211G.

If you forgot the user password but you still have access to admin password, you can login to Web Jeeves and change the user password. But, if you forgot the admin password, you need to default the Password using access code '#***' from the Phone, connected to the ATA211G.

Refer [“Reinstate Factory Defaults”](#) and [“Configuring SETU ATA211G”](#) for more details.

Q.16 I am not able to access the Web Jeeves since I forgot the IP address of LAN port of ATA211G. How can I know the IP address of LAN port?

Ans. You can use the system commands to know the IP address of LAN/WAN port of ATA211G. Ensure that you have the SLT (compatible for CLI type) connected to the FXS port.

Q.17 Can I make call to PSTN, GSM number using the Mobile port of ATA211G?

Ans. Yes. You can make call to any PSTN, GSM number using the Mobile port of ATA211G. You need to select the Mobile port by the command for 'Selective Trunk Access - Mobile', #83 followed by the number to be dialed. You can also do this by configuring the 'Outgoing Call Route' to 'Mobile' in the Web page 'FXS Port', before dialing the desired number or by setting 'Dialed Number Based Routing'.

Q.18 Incoming calls on Mobile port of ATA211G does not place on FXS port. Why?

Ans. Check the destination route configured in Webpage 'Mobile Port'.

Q.19 Can I change the configuration of ATA211G to factory default?

Ans. Yes. You can change the configuration to factory default. Refer [“Reinstate Factory Defaults”](#) for this information.

Q.20 Can I select the SIP Trunk or Mobile port to make OG call?

Ans. Yes. You can dial first the access code for the Mobile, SIP1, SIP2 or SIP3 and then dial the required number for OG call.

Q.21 Can I use the ATA211G for Least Cost Routing (LCR) application?

Ans. Yes. You can do this by selecting 'Dialed Number based Routing' for parameter 'OG Call Route' in Web page for 'FXS Port'. After selecting option for routing, program in the Webpage 'Dialed Number Table', the numbers to be dialed and corresponding SIP Trunk or Mobile port as Destination Route which is cheaper for the call to be made.

Q.22 Can I route the Incoming calls on SIP to Mobile Port?

Ans. Yes. You can configure suitable routing option for 'Route the Incoming Calls' in the Webpage 'SIP Trunk'. Refer topic [“SIP Trunks”](#).

Q.23 Though SIM card is inserted in the SIM card slot, LED labeled as 'L2' is blinking fast in Power On state?

Ans. Check whether the SIM card is inserted properly. If not, insert SIM card properly and restart the system. Also check whether the SIM is activated. If deactivated, activate the SIM and check.

Q.24 LEDs are not glowing after power up?

Ans. Check the power supply between mains and DC jack of ATA211G.

Q.25 Can I use the SIM PIN protection in ATA211G?

Ans. Yes, you can use the SIM PIN protection in ATA211G. (Refer SIM PIN feature in [“Mobile Port”](#) topic)

Q.27 ATA211G does not get registered to GSM network?

Ans. Check whether the SIM card is properly inserted and it is activated. Check SIM PIN. Check the availability of good GSM signal. Check the connectivity of antenna etc. (Refer SIM PIN feature in Mobile Port topic and 'Signal Strength' for more details)

Q.28 Incoming calls on Mobile port are not being placed on FXS Port however, caller still gets ring back tone.

Ans. Check the connectivity of analog phone to the FXS port of ATA211G.

Q.29 I am hearing garbled voice when call is made through Mobile port. What should I do?

Ans. If you are using antenna without wire, use wired antenna for better gain. If you are using wired antenna, change the antenna position.

Q.30 I have EPABX with FXS and FXO interface. Can ATA211G be integrated with existing EPABX and also reduces the call cost?

Ans. Yes. You can connect ATA211G with your existing EPABX. When connected with FXO port of EPABX, it enables you to make calls from the extensions of EPABX, directly to GSM network or VoIP which bypass the PSTN exchange and results in significant saving in call costs.

Acronyms

ASCII	American Standard Code for Information Interchange
CLI	Caller Line Identification
CPT	Call Progress Tone
CPTG	Call Progress Tone Generation
CWT	Call Waiting Tone
DHCP	Dynamic Host Configuration Protocol
DND	Do Not Disturb
DNS	Domain Name Service
DST	Daylight Savings Time
DTMF	Dual Tone Multi-Frequency
FEC	Forward Error Correction
FTP	File Transfer Protocol
GMT	Greenwich Mean Time
GSM	Global Systems for Mobile Communications
IANA	Internet Assigned Numbers Authority
IC	Incoming call
IP	Internet Protocol
ISP	Internet Service Provider
ITSP	Internet Telephony Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network
L2 VLAN	Layer2 Virtual Local Area Network
LED	Light Emitting Diode
MAC	Media Access Control
ms	Milliseconds
MPLS	Multi Protocol Label Switching
NAT	Network Address Traversal
NTP	Network Time Protocol
OG	Outgoing Call
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
P2P	Peer to Peer
PPPoE	Point to Point Protocol over Ethernet
QoS	Quality of Service

RBT	Ring Back Tone
RFC	Request for Comments
RTP	Real Time Transport Protocol
SIP	Session Initiation Protocol
SLT	Single Line Telephone
SNTP	Simple Network Time Protocol
SOHO	Small Office Home Office
STUN	Simple Traversal of UDP over NAT
TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Universal Reference/Resource Locator
VAD	Voice Activity Detection
WAN	Wide Area Network
VoIP	Voice Over Internet Protocol

Features at a Glance

Description	Feature Code
To enter programming code for Admin	#19-Admin Password
To enter programming code for User	#18-User Password
To exit programming mode	0
Speed Dialing	#9-Index Number
To default Login Password	#***
Make a new call	#81
Disconnect the call	#82
Selective Trunk Access- Mobile	#83
Selective Trunk Access-SIP1	#84
Selective Trunk Access-SIP2	#85
Selective Trunk Access-SIP3	#86

System Commands

Command	Meaning	Allowed to?
11-WAN IP Address-#*	To program the WAN IP Address	Admin and User
12-LAN IP Address-#*	To program the LAN IP Address	Admin and User
13-Subnet Mask-#*	To program the Subnet Mask of WAN port	Admin and User
14-Subnet Mask-#*	To program the Subnet Mask of LAN port	Admin and User
15-Code-#*	To program the connection type of WAN port; where, Code = 1 for DHCP, Code = 2 for PPPoE and Code=3 for Static	Admin and User
16-Code-#*	To enable/disable 'L2 VLAN/CoS' on WAN Port; where, Code=0 for Disable and Code=1 for Enable	Admin and User
21-Reverse Admin/User Password-#*	To default the ATA211G	Admin and User
31- #*	To display the WAN IP Address on SLT	Admin and User
32-#*	To display the LAN IP address on SLT	Admin and User
33-#*	To display the WAN Subnet Mask on SLT	Admin and User
34-#*	To display the LAN Subnet Mask on SLT	Admin and User
35-#*	To display the gateway address on SLT	Admin and User
36-#*	To display the Primary DNS address on SLT	Admin and User
37-#*	To display the connection type of WAN port on SLT	Admin and User
38 - SIP Trunk - #*	To display the registration status of SIP Trunk on SLT	Admin and User
39-#*	To display the Secondary DNS address on SLT	Admin and User
51-Destination Number-#*	To program Call Forward-Unconditional number	Admin and User
52-Code-#*	Set/Cancel Call Forward-Unconditional; where, Code=1 to set and Code=0 to cancel the Call Forward-Unconditional	Admin and User
53-Destination Number-#*	To program Call Forward-When Busy number	Admin and User
54-Code-#*	Set/Cancel Call Forward-When Busy; where, Code=1 to set and Code=0 to cancel Call Forward-When Busy Number	Admin and User
55-Destination Number-#*	To program Call Forward-When No Reply number	Admin and User
56-Code-#*	Set/Cancel Call Forward-When No Reply; where, Code=1 to set and Code=0 to cancel Call Forward-When No Reply	Admin and User
57-XX-#* (XX is time in seconds)	Set Call Forward Timer	Admin and User
61-Code-#*	Set/Cancel DND; where, Code=1 to set and Code=0 to cancel DND	Admin and User
66-SIP Trunk-Code-#*	Send/Block Caller ID for SIP1, SIP2, SIP3; where, SIP Trunk=1 for SIP1, SIP Trunk=2 for SIP2, SIP Trunk=3 for SIP3 and Code=1 to block, Code=0 to send Caller ID	Only Admin

Command	Meaning	Allowed to?
72-Code-#*	Set/Cancel Call Waiting; where, Code=1 to set and Code=2 to cancel Call Waiting	Admin and User
81-#*	To know the Signal Strength on phone LCD	Admin and User
82-Code-#*	To enable/disable CLIR on Mobile Port	Only Admin
99-#*	Cancel all features	Admin and User

Product Specifications

PORT NAME	APPLICATION	NO. OF PORTS	CONNECTOR TYPE
FXS Port	For connecting Analog Phone or Fax Machine	1	RJ 11
LAN Port	To connect a Computer	1	RJ 45, 10/100 Base T
WAN Port	To connect Ethernet cable router provided by service provider	1	RJ 45, 10/100 Base T
Mobile Port	To connect to GSM Network	1	Antenna
DC Jack	To connect the power adapter for DC supply voltage	1	DC Power Jack

VoIP and Networking Protocol

- SIP (RFC 3261)
- IPv4- Internet Protocol Version 4 (RFC 791)
- ARP- Address Resolution Protocol (RFC 0826)
- ICMP- Internet Control Message Protocol (RFC 0792)
- TCP- Transmission Control protocol (RFC 793)
- UDP- User Datagram Protocol (RFC 768)
- RTP- Real Time Protocol (RFC 1889)
- RTCP- Real Time Control Protocol (RFC 1889)
- SNTP- Simple Network Time Protocol
- PPP- Point-to-Point Protocol (RFC 1661)
- DNS- Protocol (RFC 2136)
- Outbound Proxy
- STUN
- NAT Keep Alive
- VLAN- Virtual LAN (Layer2)

GSM Parameters

- GSM Frequency Band: Quad-band: GSM 850, EGSM 900, DCS 1800, PCS 1900
- Compliant: ETSI GSM Phase 2/2+
- SIM Card: One SIM for GSM Port
- SIM Interface: 1.8V, 3V
- Transmission Power: Class 4 (2W) at EGSM900 and GSM850; Class 1 (1W) at DCS1800 and PCS 1900
- RF Sensitivity: Better than -106dBm

Antenna Parameters

- Type of Antenna: Fixed Omni Directional Swivel Antenna
- Antenna Gain: 2.5dBi
- Antenna Connector: SMA (Male), 50Ω Impedance

Voice Functionalities

- Three SIP Trunks
- DTMF: Inband, Outband (RFC 2833) and SIP INFO
- Call Progress Tone Generation: Select as per the country
- QoS (Layer3): RTP QoS and SIP QoS using DiffServ

- Full Duplex Audio
- Echo Cancellation: G. 168, with programmable tail length of 8 - 32 msecs.
- Forward Error Correction (FEC)
- Voice Activity Detection (VAD)
- Fax using T.38 and Pass Through
- Flash Timer Programming

Voice Codec's

G.711 (a-Law and μ -Law), G.729, G.723

Telephony features

- Voice Calls using SIP proxy and Voice calls without using SIP proxy (Peer-to-Peer Calling)
- Call Waiting
- Caller ID Blocking
- Call Forwarding
- 3-Party conference
- Call Transfer
- Call Hold
- Caller ID: Generation and Display
- Answer supervision
- Disconnect supervision
- Speed Dialing
- Do-Not-Disturb (DND)
- Polarity Reversal
- Call Toggle

Time Settings

- Synchronizing with specific Time Server

Provisioning, Administration and Maintenance

- Auto Configuration
- Programming using Web Jeeves
- Software Upgrade (Using FTP Server)
- PCAP Trace
- Programming using Phone of parameters such as:
 - LAN IP Address
 - WAN IP Address, etc

LED Indication

LED for Power

LED for FXS

LED for Mobile

LED for SIP1

LED for SIP2

Security

Password Protected Administrator and User settings

Dimension (W x H x D)

104mm x 27mm x 80mm (4.09" x 1.06" x 3.15")

Power Supply

External Adaptor: 12VDC@ 1.25Amp.

Power Consumption: 5W

Environmental

Operating Temperature: -10°C to 50°C (14°F to 122°F)

Operating Humidity: 5-95% RH, Non-Condensing

Storage Temperature: -40°C to 85°C (-40°F to 185°F)

Storage Humidity: 0-95% RH, Non-Condensing

Warranty Statement

Matrix warrants to its consumer purchaser any of its products to be free of defects in material, workmanship and performance for a period of 15 months from date of manufacturing or 12 months from the date of installation which ever is earlier. During this warranty period, Matrix will at its option, repair or replace the product at no additional charge if the product is found to have manufacturing defect. Any replacement product or part/s may be furnished on an exchange basis, which shall be new or like-new, provided that it has functionality at least equal to that of the product, being replaced. All replacement parts and products will be the property of Matrix. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only.

This limited warranty does not apply to:

1. Products that have been subjected to abuse, accident, natural disaster, misuse, modification, tampering, faulty installation, lack of reasonable care, repair or service in any way that is not contemplated in the documentation for the product, or if the model or serial number has been altered, tampered with, defaced or removed.
2. Products which have been damaged by lightning storms, water or power surges or which have been neglected, altered, used for a purpose other than the one for which they were manufactured, repaired by customer or any party without Matrix's written authorization or used in any manner inconsistent with Matrix's instructions.
3. Products received improperly packed or physically damaged.
4. Products damaged due to operation of product outside the products' specifications or use without designated protections.

Warranty valid only if:

- Primary protection on all the ports provided.
- Mains supply is within limit and protected.
- Environment conditions are maintained as per the product specifications.

Warranty Card:

When the product is installed, please return the warranty card with:

- Date, signature and stamp of the customer.
- Date, signature and stamp of the channel partner.

Matrix assumes that the customer agrees with the warranty terms even when the warranty card is not signed and returned as suggested.

The Purchaser shall have to bear shipping charges for sending product to Matrix for testing/rectification. The product shall be shipped to the Purchaser at no-charge if the material is found to be under warranty. The Purchaser shall have to either insure the product or assume liability for loss or damage during transit. Matrix reserves the right to waive off or make any changes in its warranty policy without giving any notice.

If Matrix is unable to repair or replace, as applicable, a defective product which is covered by Matrix warranty, Matrix shall, within a reasonable time after being notified of the defect, refund the purchase price of the product provided the consumer/purchaser returns the product to Matrix.

In no event will Matrix be liable for any damages including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

Except for the obligations specifically set forth in this Warranty Policy Statement, in no event shall Matrix be liable for any direct, indirect, special, incidental or consequential damages whether based on contract or any other legal theory and where advised of the possibility of such damages.

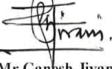
Neither Matrix nor any of its distributors, dealers or sub-dealers makes any other warranty of any kind, whether expressed or implied, with respect to Matrix products. Matrix and its distributors, dealers or sub-dealers specifically disclaim the implied warranties of merchantability and fitness for a particular purpose.

This warranty is not transferable and applies only to the original consumer purchaser of the Product. Warranty shall be void if the warranty card is not completed and registered with Matrix within 30 days of installation.

All legal course of action subjected to Vadodara (Gujarat, India) Jurisdiction only.

Regulatory Information



DECLARATION OF CONFORMITY	
Manufacturer's Name:	Matrix ComSec Pvt. Ltd
Manufacturer's Address:	38-39, GIDC, Waghodia, Dist Vadodara 391 760
Declares that the products	
Product :	Analog Telephone Adapter for VoIP Communication
Model Type:	SETU ATA211G
Trade Name:	MATRIX
Product Options	This declaration covers all options of the above products
Conforms to the following product specification.	
EMI/EMC:	
Standard	
CISPR 22 2006	
IEC 61000-3-2 2005	
IEC 61000-3-3 2005	
IEC 61000-4-2 2001	
IEC 61000-4-3 2006	
IEC 61000-4-4 2004	
IEC 61000-4-5 2005	
IEC 61000-4-6 2004	
IEC 61000-4-8 2001	
IEC 61000-4-11 2004	
SAFETY	
IEC 60950-1 2005 Second Edition.	
Supplementary Information:	
The Product herewith comply with the requirement of the Low Voltage Directive 73/23/EEC and the EMC Directive 89/336/EEC and carry the CE-marking accordingly.	
 Mr. Ganesh Jivani Director Date: 04 December 2010	 

MATRIX COMSEC PVT. LTD.
Head Office: 39-4 GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Fax: +91 265 2636598, Email: Inquiry@MatrixComSec.com, www.MatrixComSec.com
Factory: 39-GIDC, Waghodia-391 760, Dist. Vadodara, India. Ph: +91 2668 262056/57, Fax: +91 2668 262631

Declaration of Conformity (RoHS Declaration)

We, Matrix Comsec Private Limited hereby declare that the product listed below, to which this Declaration of Conformity relates, is in the conformity with the requirement of the following European Union Directive for RoHS compliance:

Document No.	Title	Edition/Date of Issue
EN 2002/95/EC	Restriction of Hazardous Substances	27 January 2003

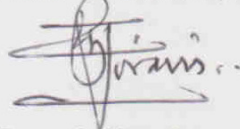
Product : Analog Telephone Adapter for VoIP Communication
Model : SETU ATA211G

Manufacturer's Name : Matrix ComSec Private Limited
Address : 39 – GIDC
Waghodia – 391 760
Dist: Vadodara, India

Additional Information:

The text of EU Directive 2002/95/EC may be found at the following website:
http://europa.eu.int/eur-lex/pri/en/oj/dat/2003/l_037/l_03720030213en00190023.pdf

Date : 08 December 2010
Place : Vadodara, India



Ganesh Jivani
(Director)
Matrix ComSec Private Limited



Open Source Licensing Terms and Conditions

- The firmware of this product also includes some of the Open-Source software released under GNU General Public License (GPL) Version 2. Terms of this license is printed in full below.

GNU GENERAL PUBLIC LICENSE
Version 2, June 1991

Copyright (C) 1989, 1991 Free Software Foundation, Inc.,
51 Franklin Street, Fifth Floor, Boston, MA 02110-1301 USA
Everyone is permitted to copy and distribute verbatim copies
of this license document, but changing it is not allowed.

Preamble

The licenses for most software are designed to take away your freedom to share and change it. By contrast, the GNU General Public License is intended to guarantee your freedom to share and change free software--to make sure the software is free for all its users. This General Public License applies to most of the Free Software Foundation's software and to any other program whose authors commit to using it. (Some other Free Software Foundation software is covered by the GNU Lesser General Public License instead.) You can apply it to your programs, too.

When we speak of free software, we are referring to freedom, not price. Our General Public Licenses are designed to make sure that you have the freedom to distribute copies of free software (and charge for this service if you wish), that you receive source code or can get it if you want it, that you can change the software or use pieces of it in new free programs; and that you know you can do these things.

To protect your rights, we need to make restrictions that forbid anyone to deny you these rights or to ask you to surrender the rights. These restrictions translate to certain responsibilities for you if you distribute copies of the software, or if you modify it.

For example, if you distribute copies of such a program, whether gratis or for a fee, you must give the recipients all the rights that you have. You must make sure that they, too, receive or can get the source code. And you must show them these terms so they know their rights.

We protect your rights with two steps: (1) copyright the software, and (2) offer you this license which gives you legal permission to copy, distribute and/or modify the software.

Also, for each author's protection and ours, we want to make certain that everyone understands that there is no warranty for this free software. If the software is modified by someone else and passed on, we want its recipients to know that what they have is not the original, so that any problems introduced by others will not reflect on the original authors' reputations.

Finally, any free program is threatened constantly by software patents. We wish to avoid the danger that redistributors of a free program will individually obtain patent licenses, in effect making the program proprietary. To prevent this, we have made it clear that any patent must be licensed for everyone's free use or not licensed at all.

The precise terms and conditions for copying, distribution and modification follow.

GNU GENERAL PUBLIC LICENSE
TERMS AND CONDITIONS FOR COPYING, DISTRIBUTION AND MODIFICATION

0. This License applies to any program or other work which contains a notice placed by the copyright holder saying it may be distributed under the terms of this General Public License. The "Program", below, refers to any such program or work, and a "work based on the Program" means either the Program or any derivative work under copyright law: that is to say, a work containing the Program or a portion of it, either verbatim or with modifications and/or translated into another language. (Hereinafter, translation is included without limitation in the term "modification".) Each licensee is addressed as "you".

Activities other than copying, distribution and modification are not covered by this License; they are outside its scope. The act of running the Program is not restricted, and the output from the Program is covered only if its contents constitute a work based on the Program (independent of having been made by running the Program). Whether that is true depends on what the Program does.

1. You may copy and distribute verbatim copies of the Program's source code as you receive it, in any medium, provided that you conspicuously and appropriately publish on each copy an appropriate copyright notice and disclaimer of warranty; keep intact all the notices that refer to this License and to the absence of any warranty; and give any other recipients of the Program a copy of this License along with the Program.

You may charge a fee for the physical act of transferring a copy, and you may at your option offer warranty protection in exchange for a fee.

2. You may modify your copy or copies of the Program or any portion

of it, thus forming a work based on the Program, and copy and distribute such modifications or work under the terms of Section 1 above, provided that you also meet all of these conditions:

- a) You must cause the modified files to carry prominent notices stating that you changed the files and the date of any change.
- b) You must cause any work that you distribute or publish, that in whole or in part contains or is derived from the Program or any part thereof, to be licensed as a whole at no charge to all third parties under the terms of this License.
- c) If the modified program normally reads commands interactively when run, you must cause it, when started running for such interactive use in the most ordinary way, to print or display an announcement including an appropriate copyright notice and a notice that there is no warranty (or else, saying that you provide a warranty) and that users may redistribute the program under these conditions, and telling the user how to view a copy of this License. (Exception: if the Program itself is interactive but does not normally print such an announcement, your work based on the Program is not required to print an announcement.)

These requirements apply to the modified work as a whole. If identifiable sections of that work are not derived from the Program, and can be reasonably considered independent and separate works in themselves, then this License, and its terms, do not apply to those sections when you distribute them as separate works. But when you distribute the same sections as part of a whole which is a work based on the Program, the distribution of the whole must be on the terms of this License, whose permissions for other licensees extend to the entire whole, and thus to each and every part regardless of who wrote it.

Thus, it is not the intent of this section to claim rights or contest your rights to work written entirely by you; rather, the intent is to exercise the right to control the distribution of derivative or collective works based on the Program.

In addition, mere aggregation of another work not based on the Program with the Program (or with a work based on the Program) on a volume of a storage or distribution medium does not bring the other work under the scope of this License.

3. You may copy and distribute the Program (or a work based on it, under Section 2) in object code or executable form under the terms of Sections 1 and 2 above provided that you also do one of the following:

- a) Accompany it with the complete corresponding machine-readable source code, which must be distributed under the terms of Sections 1 and 2 above on a medium customarily used for software interchange; or,
- b) Accompany it with a written offer, valid for at least three years, to give any third party, for a charge no more than your

cost of physically performing source distribution, a complete machine-readable copy of the corresponding source code, to be distributed under the terms of Sections 1 and 2 above on a medium customarily used for software interchange; or,

c) Accompany it with the information you received as to the offer to distribute corresponding source code. (This alternative is allowed only for noncommercial distribution and only if you received the program in object code or executable form with such an offer, in accord with Subsection b above.)

The source code for a work means the preferred form of the work for making modifications to it. For an executable work, complete source code means all the source code for all modules it contains, plus any associated interface definition files, plus the scripts used to control compilation and installation of the executable. However, as a special exception, the source code distributed need not include anything that is normally distributed (in either source or binary form) with the major components (compiler, kernel, and so on) of the operating system on which the executable runs, unless that component itself accompanies the executable.

If distribution of executable or object code is made by offering access to copy from a designated place, then offering equivalent access to copy the source code from the same place counts as distribution of the source code, even though third parties are not compelled to copy the source along with the object code.

4. You may not copy, modify, sublicense, or distribute the Program except as expressly provided under this License. Any attempt otherwise to copy, modify, sublicense or distribute the Program is void, and will automatically terminate your rights under this License. However, parties who have received copies, or rights, from you under this License will not have their licenses terminated so long as such parties remain in full compliance.

5. You are not required to accept this License, since you have not signed it. However, nothing else grants you permission to modify or distribute the Program or its derivative works. These actions are prohibited by law if you do not accept this License. Therefore, by modifying or distributing the Program (or any work based on the Program), you indicate your acceptance of this License to do so, and all its terms and conditions for copying, distributing or modifying the Program or works based on it.

6. Each time you redistribute the Program (or any work based on the Program), the recipient automatically receives a license from the original licensor to copy, distribute or modify the Program subject to these terms and conditions. You may not impose any further restrictions on the recipients' exercise of the rights granted herein. You are not responsible for enforcing compliance by third parties to this License.

7. If, as a consequence of a court judgment or allegation of patent infringement or for any other reason (not limited to patent issues), conditions are imposed on you (whether by court order, agreement or otherwise) that contradict the conditions of this License, they do not excuse you from the conditions of this License. If you cannot distribute so as to satisfy simultaneously your obligations under this License and any other pertinent obligations, then as a consequence you may not distribute the Program at all. For example, if a patent license would not permit royalty-free redistribution of the Program by all those who receive copies directly or indirectly through you, then the only way you could satisfy both it and this License would be to refrain entirely from distribution of the Program.

If any portion of this section is held invalid or unenforceable under any particular circumstance, the balance of the section is intended to apply and the section as a whole is intended to apply in other circumstances.

It is not the purpose of this section to induce you to infringe any patents or other property right claims or to contest validity of any such claims; this section has the sole purpose of protecting the integrity of the free software distribution system, which is implemented by public license practices. Many people have made generous contributions to the wide range of software distributed through that system in reliance on consistent application of that system; it is up to the author/donor to decide if he or she is willing to distribute software through any other system and a licensee cannot impose that choice.

This section is intended to make thoroughly clear what is believed to be a consequence of the rest of this License.

8. If the distribution and/or use of the Program is restricted in certain countries either by patents or by copyrighted interfaces, the original copyright holder who places the Program under this License may add an explicit geographical distribution limitation excluding those countries, so that distribution is permitted only in or among countries not thus excluded. In such case, this License incorporates the limitation as if written in the body of this License.

9. The Free Software Foundation may publish revised and/or new versions of the General Public License from time to time. Such new versions will be similar in spirit to the present version, but may differ in detail to address new problems or concerns.

Each version is given a distinguishing version number. If the Program specifies a version number of this License which applies to it and "any later version", you have the option of following the terms and conditions either of that version or of any later version published by the Free Software Foundation. If the Program does not specify a version number of this License, you may choose any version ever published by the Free Software Foundation.

10. If you wish to incorporate parts of the Program into other free programs whose distribution conditions are different, write to the author to ask for permission. For software which is copyrighted by the Free Software Foundation, write to the Free Software Foundation; we sometimes make exceptions for this. Our decision will be guided by the two goals of preserving the free status of all derivatives of our free software and of promoting the sharing and reuse of software generally.

NO WARRANTY

11. BECAUSE THE PROGRAM IS LICENSED FREE OF CHARGE, THERE IS NO WARRANTY FOR THE PROGRAM, TO THE EXTENT PERMITTED BY APPLICABLE LAW. EXCEPT WHEN OTHERWISE STATED IN WRITING THE COPYRIGHT HOLDERS AND/OR OTHER PARTIES PROVIDE THE PROGRAM "AS IS" WITHOUT WARRANTY OF ANY KIND, EITHER EXPRESSED OR IMPLIED, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. THE ENTIRE RISK AS TO THE QUALITY AND PERFORMANCE OF THE PROGRAM IS WITH YOU. SHOULD THE PROGRAM PROVE DEFECTIVE, YOU ASSUME THE COST OF ALL NECESSARY SERVICING, REPAIR OR CORRECTION.

12. IN NO EVENT UNLESS REQUIRED BY APPLICABLE LAW OR AGREED TO IN WRITING WILL ANY COPYRIGHT HOLDER, OR ANY OTHER PARTY WHO MAY MODIFY AND/OR REDISTRIBUTE THE PROGRAM AS PERMITTED ABOVE, BE LIABLE TO YOU FOR DAMAGES, INCLUDING ANY GENERAL, SPECIAL, INCIDENTAL OR CONSEQUENTIAL DAMAGES ARISING OUT OF THE USE OR INABILITY TO USE THE PROGRAM (INCLUDING BUT NOT LIMITED TO LOSS OF DATA OR DATA BEING RENDERED INACCURATE OR LOSSES SUSTAINED BY YOU OR THIRD PARTIES OR A FAILURE OF THE PROGRAM TO OPERATE WITH ANY OTHER PROGRAMS), EVEN IF SUCH HOLDER OR OTHER PARTY HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

END OF TERMS AND CONDITIONS

How to Apply These Terms to Your New Programs

If you develop a new program, and you want it to be of the greatest possible use to the public, the best way to achieve this is to make it free software which everyone can redistribute and change under these terms.

To do so, attach the following notices to the program. It is safest to attach them to the start of each source file to most effectively convey the exclusion of warranty; and each file should have at least the "copyright" line and a pointer to where the full notice is found.

```
<one line to give the program's name and a brief idea of what it does.>
Copyright (C) <year> <name of author>
```

```
This program is free software; you can redistribute it and/or modify
it under the terms of the GNU General Public License as published by
the Free Software Foundation; either version 2 of the License, or
(at your option) any later version.
```

```
This program is distributed in the hope that it will be useful,
but WITHOUT ANY WARRANTY; without even the implied warranty of
```

MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE. See the GNU General Public License for more details.

You should have received a copy of the GNU General Public License along with this program; if not, write to the Free Software Foundation, Inc., 51 Franklin Street, Fifth Floor, Boston, MA 02110-1301 USA.

Also add information on how to contact you by electronic and paper mail.

If the program is interactive, make it output a short notice like this when it starts in an interactive mode:

```
Gnomovision version 69, Copyright (C) year name of author
Gnomovision comes with ABSOLUTELY NO WARRANTY; for details type `show w'.
This is free software, and you are welcome to redistribute it
under certain conditions; type `show c' for details.
```

The hypothetical commands `show w' and `show c' should show the appropriate parts of the General Public License. Of course, the commands you use may be called something other than `show w' and `show c'; they could even be mouse-clicks or menu items--whatever suits your program.

You should also get your employer (if you work as a programmer) or your school, if any, to sign a "copyright disclaimer" for the program, if necessary. Here is a sample; alter the names:

```
Yoyodyne, Inc., hereby disclaims all copyright interest in the program
`Gnomovision' (which makes passes at compilers) written by James Hacker.
```

```
<signature of Ty Coon>, 1 April 1989
Ty Coon, President of Vice
```

This General Public License does not permit incorporating your program into proprietary programs. If your program is a subroutine library, you may consider it more useful to permit linking proprietary applications with the library. If this is what you want to do, use the GNU Lesser General Public License instead of this License.

Index

A

AC Impedance 54

Admin 16

Answer Signaling 53

Automatic Number Translation 44

C

Call Hold 105

Call Release Timer 65

Call Toggle 109

Call Transfer 107

Call Transfer - Attended 107

Call Transfer - Blind 107

Call Transfer-Attended 107

Call Transfer-Blind 107

Class of Service 52, 61

Cloned 62

Connection Type

DHCP 60

PPPoE 60

Static 60

Current Date 148

Current Time 148

D

Daylight Saving Time 85

Debug

configure system debug 95

debugs 95

Dialed Number Table 71

Destination Route 41, 50, 70, 73

Maximum Digits 41, 50, 70, 73, 41, 50, 70, 73

Number 27, 41, 50, 70, 73, 76

DiffServe 62

Digest Authentication 80

Disconnect Signaling 53

DNS Setting 61

DNS Address 61

Static DNS 61

E

Echo Cancellation 54

End of Dialing 136

G

General Request Timer 68

L

LEDs 14

M

Mobile Emergency Numbers 78

Mobile Network Selection 45

N

NAT Type 36

P

PCAP 97

prioritization of traffic 61

R

Receive Gain 46

Registration Retry Timer 29

Reinstate Factory Defaults 91, 94

Re-registration Timer 29

Return Call to Original Caller 139

Ring 23

rinstance 29

Routing Mode 59

Routing Tone 152

RTP QoS 62

S

SIM PIN 46

SIP ID 26, 28

SIP INVITE Timer 68

SIP Over TCP 67
SIP Provisional Timer 68
SIP Trunk Mode 26
Speed Dialing 123
Static Routing 88
Syslogs 95

T

TCP NAT Keep Alive
Interval 67
Three-party Conference 113
Time Offset 85
Transmit Gain 46

U

UDP NAT Keep Alive
Interval 67
Unique 62
User 16

V

VLAN header 61
VLAN ID 61
VLAN/CoS Layer
RTP CoS 62
SIP CoS 61
Vocoders 33
Voice Activity Detection (VAD) 65
Volume 55



Magyarországon a Matrix Telecom Ltd. képviselete,
Matrix termékek importőre, kizárólagos forgalmazója:



1095 Budapest, Mester u. 34.

Telefon: *218-5542, 215-9771, 215-7550, 216-7017, 216-7018
Fax: 218-5542 Mobil: 30 940-1970, 20 949-2688
E-mail: delton@delton.hu Web: www.delton.hu